MITEL

COMMUNICATIONS DIRECTOR

VOICE QUALITY TROUBLESHOOTING GUIDE
MITEL COMMUNICATIONS DIRECTOR RELEASE 6.0

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Voice Quality Troubleshooting Guide
Mitel Communications Director (MCD) Release 6.0
Rev. A

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Introduction

Purpose

This guide provides information on how to troubleshoot voice quality issues on the Mitel Communications Director (MCD) platform and its supported applications.

The focus of this guide is on practical troubleshooting. It does not describe network issues in detail, nor does it describe network concepts, considerations, or engineering rules . For detailed network requirements and for information how to configure a network to ensure VoIP quality, refer to the *MCD Engineering Guidelines*.

Audience

This guide is intended for use by Mitel certified MCD technicians and Mitel Product Support. Technicians who use this guide should be familiar with

- LAN equipment: for example, layer 2 switches and routers
- Analog telephony
- VoIP telephony
- MCD Engineering Guidelines.

Causes of Voice Quality Issues

Voice quality issues are present when phone users complain that they are experiencing audio degradation on calls. For example, users may complain of echo, buzzing, ticking, distortion, choppy audio, loudness issues, or delayed speech. The cause of voice quality problems can be associated with the TDM network, analog network, PSTN, IP network, user's environment, or due to configuration errors in the equipment:

- Analog, TDM, or PSTN Networks
  - Electrical Echo
  - Positive Feedback
- IP Network
  - Delay
  - Jitter
  - Speech Compression
  - Packet Loss
  - Lack of WAN Bandwidth
  - Lack of LAN Bandwidth
  - Excessive number of transcoding hops

- Environment
  - Acoustic Echo
  - Crosstalk
  - Background Noise
  - Side Tone (loud, quiet, or missing)
  - Voltage Interference
  - Improper Grounding
  - Improper Use of Handset (breathing into microphone)
- Configuration Errors
  - MCD Platforms or Gateways
  - IP phones
  - Layer 2 Ethernet Switches or routers

If you can answer the following questions, it will be easier to diagnose and correct the problem:

- What is the issue or complaint?
- What does it sound like? Do both parties on the call have the same complaint? If not, which call party (called, calling, Mitel user, external) is experiencing the voice quality issue?
- When does it occur?
- What is the call situation? For example, does it occur only on Loop Start to IP conference calls? Does it always occur at a certain time of day or at a specific phone?
- When did it start happening? For example, after a software upgrade, after new LAN equipment was added, after there where changes in the IP network, or after a new carrier was added?
- Where does the problem originate? For example, does it originate in Mitel equipment or in another vendor's customer premise equipment?
- Where is the problem occurring? For example, does it occur at a single IP phone, at multiple IP phones on the same layer 2 switch, on analog phones connected to an Analog Service Unit, on a specific trunk, or on all trunks of a specific type?

The checklists in this guide will help you collect the problem details.

Terminology

Mitel uses the following voice quality terminology:

Table 1: Voice Quality Terminology

Term	Definition
Automatic Route Selection (ARS)	ARS is a function of the MCD system that simplifies local and long distance dialing by automatically selecting the most convenient and cost-effective route, and by inserting and/or deleting digits for proper routing. An alternative name is Call Route Selection. You program ARS in the MCD System Administration Tool. You need to understand ARS programming when troubleshooting voice quality issues on trunk calls, because the route determines the system equipment that is used to establish the call. You can then focus your troubleshooting efforts on the equipment in the call path.
Analog Service Unit (ASU)	An optional Mitel Networks hardware unit that connects analog sets and analog trunks to the system.
Bandwidth	Amount of data per unit time that can be transmitted over a network connection. Typically, bandwidth is expressed in Kbps or Mbps.
Basic Rate Interface (BRI)	ISDN standards and specifications for provision of low-speed ISDN services. This standard supports two B-channels of 64 Kbps each and one D-channel of 16 Kbps on a single wire pair. BRI supports the simultaneous digital transmission of voice, video, data, and other network services over the traditional circuits of the public switched telephone network.
Carrier Sense Multiple Access / Collision Detection (CSMA/CD)	CSMA/CD is a contention protocol that determines how data is transmitted on a network. With this protocol, devices compete with each other for the available bandwidth. When data transmissions collide, the devices are notified and they simply re-send the data.
Central Office Equipment (COE)	The switching and amplifying equipment in the central office.
Central Office (CO)	The local telephone company office which connects to all local loops in a given area and where circuit switching of customer lines occurs. The local telephone company office can also be referred to as the telephone exchange.
Coder/Decoder (CODEC)	Device that converts audio (or video) from one transmission format to another. CODECs convert the following formats: analog, 16-bit linear, G.711 (u-law or A law), G729a, and G.722.1
Compression	See Speech Compression.
Crosstalk	Crosstalk occurs when an audio signal is improperly coupled from one channel to another. The most common and noticeable form of crosstalk is between channels, where a different conversation or signaling tones from another channel can be heard. Crosstalk can occur in handset cables and sound similar to echo. Crosstalk in handset cables is normally small and will be cancelled by an echo suppressor or echo canceller.

Table 1: Voice Quality Terminology

Term	Definition
Customer Premise Equipment (CPE)	The telephone equipment on the customer site. On a Mitel customer site, the MCD system is the CPE. Incoming calls that arrive at the MCD over the PSTN typically originate at CPE equipment located on the other side of the PSTN.
Digital Service Processor (DSP)	A specialized microprocessor that optimizes the processing of signals such as audio or video.
Double Talk	Occurs when more than one speaker is talking at the same time.
Echo (acoustic)	When acoustic echo is present on a call, one or both parties hear what they said tens or hundreds of milliseconds later. acoustic echo originates at the phone. Acoustic echo is caused by sound from the speaker being reflected back into the handset microphone or by a user’s voice being reflected off nearby surfaces and picked up by the microphone. If the round-trip delay of the reflection is less than 50 ms, these reflections result in sidetone (*see “Sidetone” on page 7*). However, if the round-trip delay is greater than 50 ms, these reflections can result in echo. Echo will be heard by the far-end user if the echo suppressor or canceller in the phone fails to remove all acoustic or mechanical reflection, or if the required echo suppressor or echo canceller is not present.
Echo (electrical)	When electrical echo is present on a call, one or both parties hear what they said a few tens or hundreds of milliseconds later. The reflections that cause electrical echo originate at line hybrids. These reflections cause echo if the round trip delay is long (typically greater than 50 ms) and the echo is not removed by an echo canceller.
Echo (line)	Line echo is another term for electrical echo.
Echo cancellers	A device that removes echo from a signal by adaptively predicting the echo signal and then subtracting the predicted echo from the signal Since the mathematical model used to predict the amount of echo to be removed is not an exact model and since there are always some non-linearities in the echo path, not all echo will be removed. An echo canceller normally includes an echo suppressor, also called a non-linear processor (NLP), to remove this residual echo.
Echo suppressor	A device, often called a non-linear processor, that detects the presence of echo and blocks or suppresses the signal that contains the echo. As a side effect, echo suppression can potentially block a portion of the valid signal. An echo suppressor is effective at removing low levels of echo without a significant effect on audio quality.
Handset Audio Tap	A device for recording the audio signal that is heard by a phone listener. You can use the recording to check for voice quality issues, such as residual echo in an echo canceller or acoustic echo from a handset.
Hub	A hardware device that connects multiple twisted pair or fiber optic Ethernet devices together and makes them act as a single network segment. Hubs work at the physical layer (layer 1) of the OSI model. Mitel IP phones should not be connected to the network through hubs.

Table 1: Voice Quality Terminology

Term	Definition
Hybrids (Line)	Hybrids are components (typically line cards or circuit modules) that adapt the four-wire digital network to the two-wire analog network used in the PSTN.
Jitter (packet arrival jitter)	The difference between the minimum amount of time and the maximum amount of time required for a packet to propagate through the packet network, plus any jitter in the transmission of the packet. Jitter can be considered the variation in latency.
Jitter buffer	A buffer used to store packets to compensate for variation in packet arrival. To ensure that packets are not lost due to late arrival, the jitter buffer must be larger than the jitter. The amount of time a packet is held in the jitter buffer contributes to the audio latency.
Latency (in calls)	Latency in a call is the amount of time it takes the caller's voice to reach the other end of the connection. As latency increases in a conversation, it becomes increasingly difficult for users to sustain a normal two-way conversation For VoIP to VoIP calls, latency includes the packet delay, the jitter buffer delay (which must be greater than the actual amount of jitter), the packetization delay, plus a number of other, usually smaller, delays associated with the system architecture.
Latency (in data network)	Latency in the data network is the time it takes for a packet of data to get from one designated point to another.
Layer 2 Switch	A network switch that forwards data traffic based on media access control (MAC) addresses.
Line Measurement Tool	A form in the System Administration Tool that allows you to determine the correct line settings for Loop Start (LS) trunks that are connected to the MCD system (AX Controller Card Chassis, Analog Main Board, Analog Option Board, or Analog Service Unit). You can eliminate most trunk-related echo problems by applying the correct line settings in the MCD trunk forms.
Local Area Network (LAN)	A computer network that spans a relatively small area, typically confined to a single building or group of buildings. LANs are often connected to other LANs. A system of interconnected LANs is called a wide-area network (WAN).
Network Service Unit (NSU)	A Mitel hardware unit that connects T1 or E1 trunks to the Mitel system.
Network Traffic Analyzer	A computer application or hardware device that allows you to • capture data on a live network connection • read data from a previously captured file that contains recorded packets. • read live data from the network • browse and filter displayed network data • detect VoIP calls in the captured data • verify Quality of Service settings Also known as a packet analyzer, protocol analyzer, or sniffer.
Non-linear Processor (NLP)	An echo suppressor that is normally part of the echo canceller. See "Echo suppressor" on page 4.

Table 1: Voice Quality Terminology

Term	Definition
On Premise Station (ONS)	Stations which are installed on the same premises as the PBX, or which can operate satisfactorily with the system when installed in adjacent premises without special circuit arrangements.
Packet Delay	The amount of time required for a packet to traverse the network. Packet delay can be measured by a ping test.
Packet Loss	The loss of data fragments (packets) during transmission over an IP network. In a VoIP network, a packet is considered lost if it never arrives at its destination or if it arrives too late to be used. When packet loss occurs during a VoIP call, tiny fragments of the conversation are lost. This issue can result in choppy voice quality.
Ping Test	A utility that you can use to determine the round-trip packet delay between a PC and a Mitel IP phone on the network.
Public Switched Telephone Network (PSTN)	The world-wide, public, circuit-switched telephone network. It consists of telephone lines, fiber optic cables, microwave transmission links, cellular networks, communications satellites, and undersea telephone cables, all inter-connected by switching centers, allowing any telephone in the world to communicate with any other.
Quality of Service (QoS) in the analog and TDM network	The performance of a communications channel or system is usually expressed in terms of QoS. The QoS will relate to the type of system, SNR (Signal to Noise Ratio), BER (Bit Error Ratio), maximum and mean throughput rate, reliability, priority, and other factors specific to each service.
Quality of Service (QoS) in the IP network	Quality of service is the audio quality experienced by an IP phone user while on a call. It is not the network equipment settings. However, certain network equipment configurations can help ensure that users receive adequate QoS. These configurations include: - IEEE 802.1p/Q (802.1Q VLAN now included as part of 802.1d): This is also known as VLAN tagging, priority, or COS (different from the PBX/telecom Class of Service). IEEE 802.1p/Q operates at Layer 2 to ensure the highest priority for voice traffic. - DiffServ (also known as DSCP): DiffServ is a fixed field in the Layer 3 information that is also used to define different service categories through TOS, priority, and precedence. DiffServ and Type of Service are similar. The older Type-of-Service values are compatible with the newer DiffServ values.
Reflection	In this document, reflection is when a portion of a signal is sent back towards the signal's source. If the round-trip delay in the reflections is short (less than 50 ms), the reflections are perceived by users as sidetone. However, if the round trip delay is long (greater than 50 ms) the reflections are heard as echo. Reflection can be caused by line hybrids or by acoustic or mechanical audio coupling at the phone.

Table 1: Voice Quality Terminology

Term	Definition
Sidetone	Sidetone is an electrical reflection that has a very short delay. Often, it is purposely transmitted from the handset microphone to the handset speaker to prevent the phone from sounding dead. On 2-wire analog phones, sidetone is a result of the local hybrid. On IP phones, special circuitry or software is usually required to create sidetone. The same reflected signal that creates the sidetone that reassures a user that the phone is not dead will generate annoying echo if the round-trip delay from the mouthpiece to the ear piece increases to more than 50 ms.
Speech Compression	Speech compression is performed by a CODEC (coder/decoder). A codec compresses speech by changing the format of the speech signal. It reduces the number of bits required to represent the speech signal. Although speech compression results in some loss of speech quality, it is usually not enough to be readily noticed by the user.
Subnet	Found in larger networks, these smaller networks are used to simplify addressing between numerous computers. Subnets connect to the central network through a router, hub, or gateway. A wireless LAN typically uses the same subnet to communication with all of its local computers. Subnetting is a technique used by a network administrator to subdivide a network into smaller networks using the same network number assignment. The reasons for subnetting also include: • Simplified administration: subnets can be managed more independently and efficiently. • Restructuring the internal network without affecting external networks • Improved security: separate internal networks are not visible to external networks. • Isolation of network traffic: subnetting can minimize network traffic.
Time Division Multiplexing (TDM)	A type of digital multiplexing that involves the combination of numerous signals for transmission on a single communications channel or line, with each signal being broken into different segments of short duration. Two or more signals are transferred apparently simultaneously as sub-channels in one communication channel, but are physically taking turns on the channel.
Voice quality issue	Voice quality on phone calls is subjective and varies from user to user depending on his or her expectations. In practical terms, a voice quality issue exists if customers report that the audio quality is poor and they are having difficulty communicating on calls. Note that in the context of this guide, *no audio* is not considered a voice quality issue because there is complete absence of voice. Refer to the *MCD Troubleshooting Guide* for situations in which there is no audio.
Wide Area Network (WAN)	A communication network that links across metropolitan, regional, or national boundaries. Businesses and governments use WANs to exchange data between various geographical locations. WANs interconnect local area networks (LANs).

Table 1: Voice Quality Terminology

Term	Definition
Wideband	A wideband signal contains a larger range of frequencies than have traditionally been used to represent a signal. At Mitel, a wideband signal is one that is sampled at 16,000 samples per second and supports a frequency range of 100 Hz to 7200 Hz. This signal is wideband relative to traditional telephony signals which use a signal sampling rate of 8000 samples per second, capable of representing a frequency range of 300 Hz to 3400 Hz.
Wireshark®	A network traffic analyzer tool available from the Wireshark Foundation.

Analog, TDM, or PSTN Network Issues

Echo

What is Echo?

When echo is present on a call, one or both parties hear what they said tens or hundreds of milliseconds later. Users find echo annoying, particularly if it is loud or if the echo delay is long.

Echo occurs on a call when a portion of a signal is sent back towards the signal's source and there is significant delay in the call path. If the round-trip delay in the reflections is greater than 50 ms these reflections are perceived by users as echo. Reflection can be caused by line hybrids or by acoustic or mechanical audio coupling at the phone.

If reflection is present, but the round trip delay is very short (less than 50 ms), the user will perceive the reflection as sidetone. Sidetone is reflection that has a very short delay. Sidetone is not a voice quality issue and is in fact beneficial because it has the effect of reassuring the user that the phone is not dead.

Echo: Electrical or Acoustic

Electrical Echo

In the case of electrical echo, the reflection is caused by changes in impedance in the line. These changes in impedance cause energy from the signal to be reflected back to the source.

Electrical reflection usually occurs at hybrids where 2-wire analog is converted to audio 4-wire analog audio (2W/4W) at analog trunks and stations. Hybrids are not applicable to T1, PRI, or SIP trunks.

This is not applicable to T1 or PRI or SIP. If the signal is delayed significantly, then the reflection is heard as echo. In general, a hybrid is always located at both ends of a 2-wire circuit whenever it connects to any Central Office or Customer Premise equipment. You can locate hybrids by identifying the location of all 2-wire circuits. Loop Start, Ground Start, DID, and 2-wire E&M trunks are examples of trunks that have a hybrid at both ends.

Note: Digital connections (PRI, BRI, T1, Mitel Digital sets and IP sets) are always 4-wire.

Two-wire analog audio refers to two physical wires (normally a twisted pair) where both the transmit (TX) and the receive (RX) audio are carried over the same physical pair of wires. Four wire analog audio refers to four physical wires where the TX and RX audio are separated from each other (that is, the TX audio is carried over a separate pair of wires from the RX audio). The device that converts the 2-wire analog audio to 4-wire analog audio is called a "hybrid" (see Figure 1 below). Although hybrids are designed to minimize reflection, they cannot completely eliminate it.

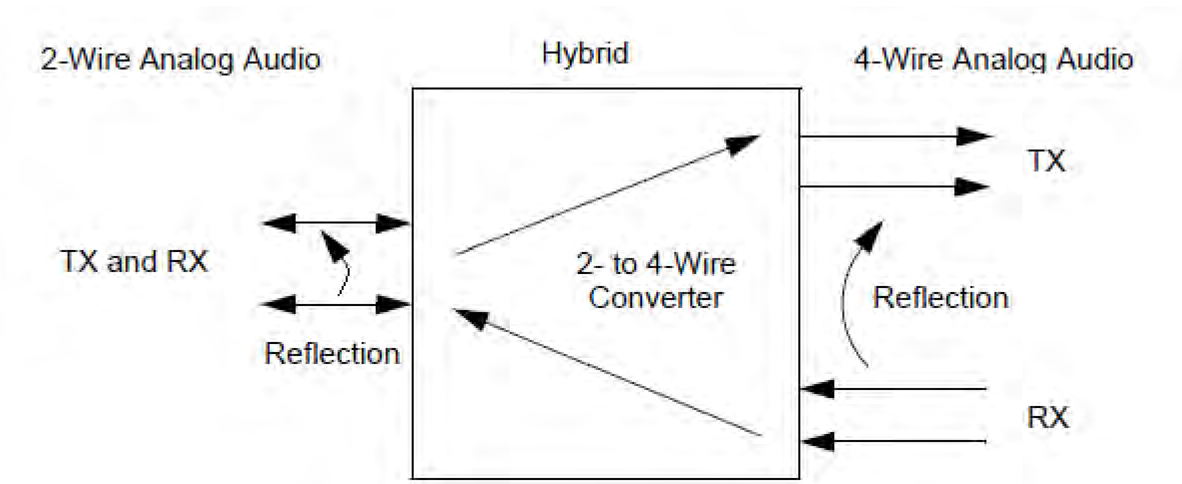

Figure 1: Reflection at a Hybrid

Another possible, but less likely, cause of electrical reflection is crosstalk between transmit and receive analog audio wires. Crosstalk can originate in faulty telephone wiring at endpoints that use separate transmit and receive analog wires. Crosstalk is one of the factors that affects xDSL speeds.

Acoustic Echo

In the case of acoustic echo, the reflection is the result of sound from the telephone earpiece or handsfree speaker being picked up by the phone's microphone. The party on the far end hears both the caller's voice and, a few tens to hundreds of milliseconds later, the reflected version of his or her own voice. Acoustic echo is more likely to occur with

- Handsfree car phone systems
- Speakerphones or phones in handsfree mode
- Standalone conference phones
- Improperly designed phones where vibrations from the loudspeaker transfer to the microphone via the handset casing.

Differences Between Sidetone and Echo

Sidetone is a feature that helps phone users moderate the volume of their voice so they don't speak too loudly or too softly to the called party. The phone system feeds some of the user's voice back though the receiver to help them regulate the volume of their voice. Callers who do not receive sidetone may consider the phone line dead. Too little sidetone will convince callers that they're not being heard and cause them to shout. Too much sidetone can cause callers to lower their voices and not be heard well at the other end of the line.

When you talk, the sound waves bounce off the walls in the room and reflect back to you, and you hear yourself. You hear this reflection as sidetone and consider it normal. That is because the speed of sound waves through air is about one millisecond per foot (1 ms/ft). In a normal size room, the round trip delay from mouth-to-wall-to-ear is less than 50 ms and you hear this as sidetone. If you are in a large auditorium or gym, the round trip delay from

mouth-to-wall-to-ear can be well over 50 ms, and you hear this reflection as echo. Phone calls involving reflections also have round trip delays. If the round trip delay of a reflection from your phone to a hybrid and back to your phone is short (for example, under 50 ms), you will hear this as sidetone, not as echo. But if the round trip delay is significant (greater than 50 ms) you will hear the reflection as echo. Table 2 below shows the differences of sidetone and echo.

Table 2: Sidetone Versus Echo

Echo	Sidetone
Reflection + Significant Delay	Reflection + Short Delay
Round trip delay >50 ms	Round trip delay < 50 ms

Where Does Electrical Echo Originate?

Electrical echo usually originates in the Customer Premise Equipment (CPE) on either side of the PSTN. The VoIP network on the Customer Premise is the long-delay segment of the call. The inherent delays in the VoIP packaging and transmission over an IP network in combination with reflection that occurs at hybrids can result in electrical echo.

It is important to note that the source of echo is not necessarily where it is being heard. Echo that originates in the CPE on one side of the PSTN might be heard in the CPE on the other end. You must educate your customer so that they understand that although echo is being heard on a Mitel phone, the echo may in fact originate in another vendor’s CPE on the other side of the PSTN.

Figure 2 illustrates how echo can originate in CPE on either site of the PSTN:

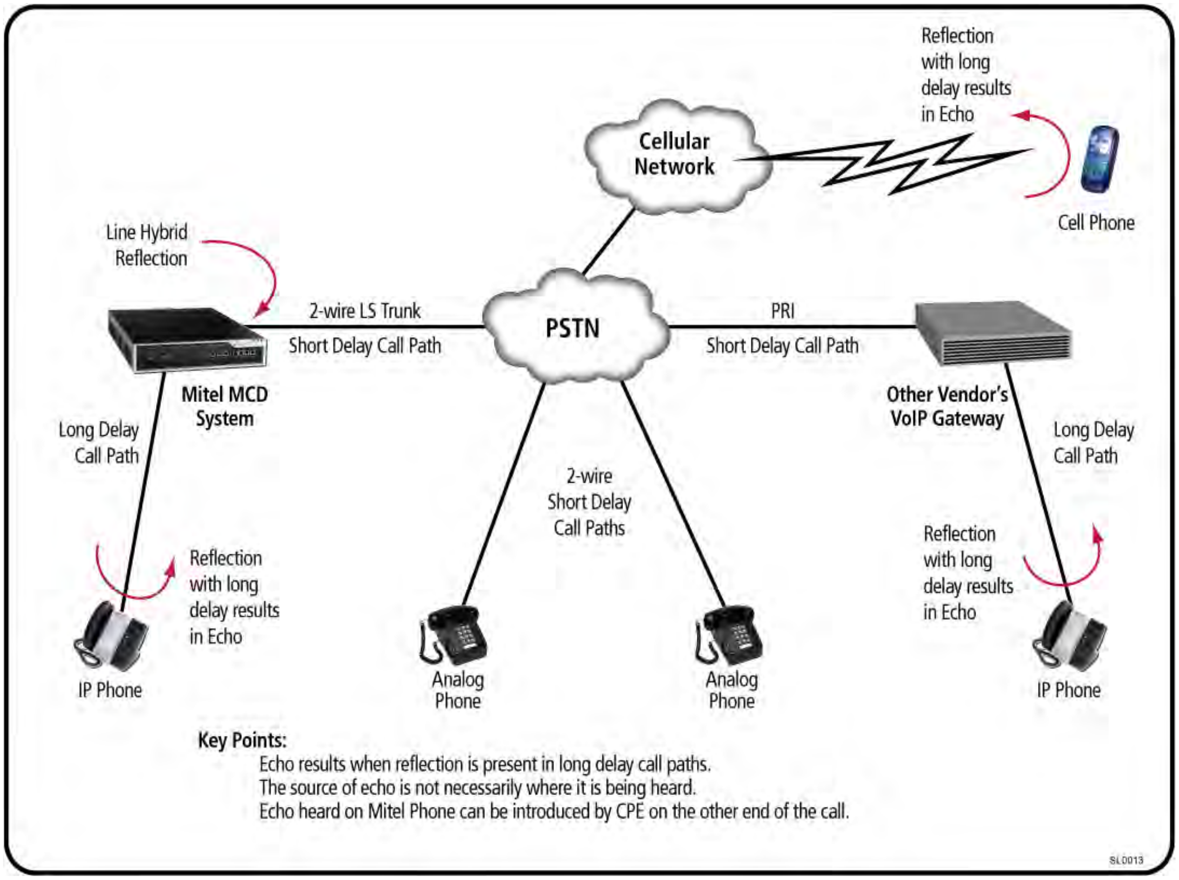

Figure 2: Echo Originating in Customer Premise Equipment

Echo in CPE can also be caused by reflections that originate in the Central Office Equipment of the Telco. Typically, reflection on the PSTN does not result in echo on PSTN calls because the call delay is usually too short. PSTN calls are normally transmitted via copper wire, optical cable, and microwave and the speed at which these signals travel is close to the speed of light (about one foot per nanosecond, or about five microseconds per mile). Signals over copper wire travel at about 1/4 to 1/2 the speed of light, so even for a long distance land-line phone call spanning 1,000 miles, the round trip delay will be less than 20 ms, and you do not hear echo.

Who is Responsible for Correcting Echo?

Any equipment that connects to an audio path with a round trip delay sufficiently long enough (> 50 ms) to cause reflections that are heard as echo must prevent (cancel) the echo from entering the long delay path. This rule is necessary because once a signal with echo has passed though a long delay path, it is usually difficult or impossible to cancel. If a reflection occurs in equipment that does not insert a long delay, that equipment is not responsible for removing the reflection. Only when the reflection is about to enter a long delay path does it need to be removed.

IP networks and cellular networks are long delay paths. Long delay paths seldom occur in the TDM network. However, inter-continental calls or calls that span the USA or Canada can sometimes have delay.

The owner of the customer premise equipment is responsible for removing any long delay reflections from entering the PSTN. This is equivalent to saying that the owner of the customer premise equipment in the long delay audio path must suppress or cancel echo that enters the long delay audio path, since it is normally difficult or impossible to cancel a reflection once it has traversed a long delay path.

- You are responsible for suppressing or cancelling long delay reflections that originate in the Mitel system from entering the PSTN.

Note: If the Mitel MCD connects to the PSTN through a 2-wire analog connection, the Mitel equipment will contain a hybrid that reflects back into the PSTN. The Mitel equipment is not responsible for cancelling this reflection.

- The owners of the customer premise equipment on the other side of the PSTN are responsible for suppressing or cancelling any echo that originates in their system from entering the PSTN. Often you can determine who is responsible for causing the echo using a simple test (see “Identifying the Source of Echo” on page 55).

The following sections illustrate some specific scenarios and identify

- the origin of the echo, and
- who is responsible for correcting the echo

Scenario 1

In Scenario 1, a Mitel IP phone is connected via IP to the MCD platform which is connected to the PSTN via 2-wire, Loop Start (LS) trunk.

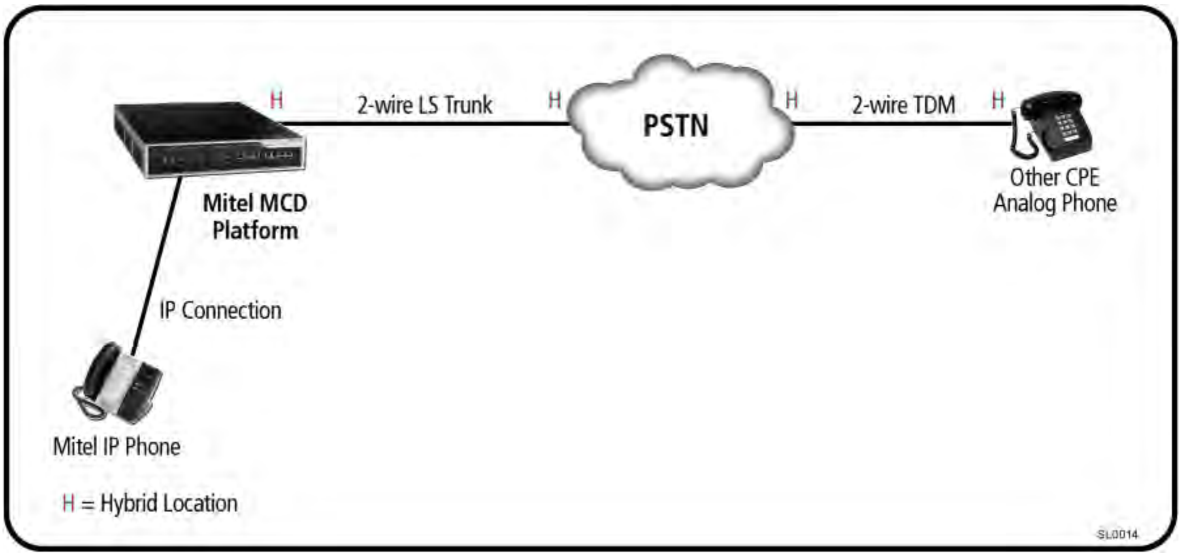

Figure 3: Scenario 1

In Scenario 1

1. With respect to reflection and echo, the PSTN treats the MCD LS trunk like a 2-wire analog set. The hybrid on the Mitel LS trunk card generates a reflection into the PSTN network just like reflection created at the the hybrid in the 2-wire PSTN set.
2. The IP path on the MCD introduces a long delay that in combination with reflection from the PSTN towards the IP phone generates echo that can heard by the IP phone user. The MCD is therefore responsible for cancelling echo from the PSTN that would otherwise be heard by the IP phone user.
3. There are four hybrids, two for each 2-wire segment, each of which contributes to the echo towards the IP phone. The echo canceller in the MCD is responsible for cancelling reflections from each of these four sources.
4. Each of the four hybrids also generate reflections in the direction of the 2-wire analog phone on the PSTN. However, the delay of these reflections is typically short enough that these reflections are all heard as sidetone.
5. If the PSTN introduces a delay that is long enough for the reflections to sound like echo, then the PSTN is responsible for placing echo cancellers in the audio path. The PSTN will normally insert a pair of echo cancellers, one at each end. These echo cancellers will prevent the 2-wire PSTN set from hearing echo, but since the MCD must have its own echo canceller to cancel echoes from the PSTN, these echo cancellers should not be expected to improve the audio quality for the Mitel IP phone user.
6. Other than the hybrid in the LS trunk card, there are no sources of electrical echo in the Mitel customer premise equipment because the internal path is all 4-wire.

7. The IP phone may generate acoustic, mechanical, or crosstalk echo. The acoustic echo suppressor or acoustic echo canceller on the handset is responsible for cancelling this echo. If the 2-wire PSTN handset user hears acoustic echo, it probably originates in the IP phone set. The IP phone is responsible for suppressing or cancelling the acoustic echo.
8. The customer premise equipment, in this case the MCD, that inserts the echo causing delays is responsible for cancelling any reflections that would be heard as echo.

Scenario 2

In scenario 2, an analog phone is connected to the MCD platform via a 2-wire ONS line; the MCD platform is connected to the PSTN via 2-wire LS trunk.

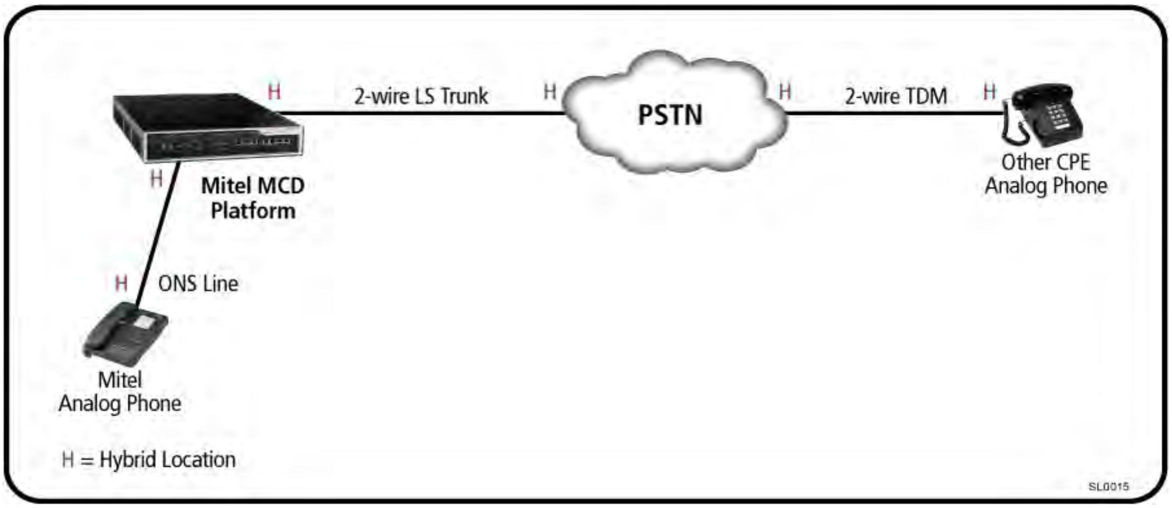

Figure 4: Scenario 2

In Scenario 2:

1. With respect to reflection and echo, the PSTN treats the MCD LS trunk like a 2-wire analog phone. The hybrid on the Mitel LS trunk card generates a reflection into the PSTN network just like the reflection created at the hybrid in the 2-wire PSTN phone.
2. The ONS line on the MCD platform does not introduce any significant delay that could cause echo. Therefore, you are not responsible for cancelling reflections from entering the PSTN for the benefit of the CPE 2-wire analog phone on the other side of the PSTN.
3. There are six hybrids, two for each 2- wire segment, each of which contributes reflections towards the Mitel 2-wire analog phone. The delays through the network are normally so short that these reflections are heard as sidetone and the MCD platform does not need remove these reflections.
4. These six hybrids also each generate a reflection in the direction of the 2-wire analog phone on the other side of the PSTN. Again, the delay of these reflections is typically short enough that these reflections are all heard as sidetone.
5. If the PSTN introduces a delay that is long enough for the reflections to result in echo, then the PSTN is responsible for placing echo cancellers in the audio path. In this case, the PSTN will normally insert a pair of echo cancellers, one at each end. These echo cancellers

prevent both the 2-wire PSTN analog phone and the 2-wire Mitel analog phone from hearing echo.

6. There are three hybrids in the Mitel CPE that cause reflections in the direction of the PSTN 2-wire analog phone. Since the delay though the MCD system is short, these three reflections cannot be distinguished from a single reflection and can be handled by the PSTN as if they originated from a single two-wire phone hybrid.
7. The 2-wire analog phone may generate acoustic, mechanical, or crosstalk reflections. Normally the levels of these reflections are much lower than the hybrid reflection, with approximately the same delay, so no acoustic echo canceller or suppressor is needed. However, if the 2-wire set has a handsfree mode, echo suppression will be needed because of the louder levels and the additional delays caused by longer acoustic echo paths.
8. If Mitel equipment inserts long delays that cause echo, you are responsible for cancelling the reflections that cause the echo. Delayed reflections that cause echo may occasionally be caused by handsfree acoustic echo or long PSTN delays.

Scenario 3

In scenario 3, a Mitel IP phone is connected via IP to the MCD platform; the MCD is connected to the PSTN via PRI trunk; the PSTN is connected to a VoIP Gateway via LS trunk; and a VoIP phone is connected via IP to the gateway.

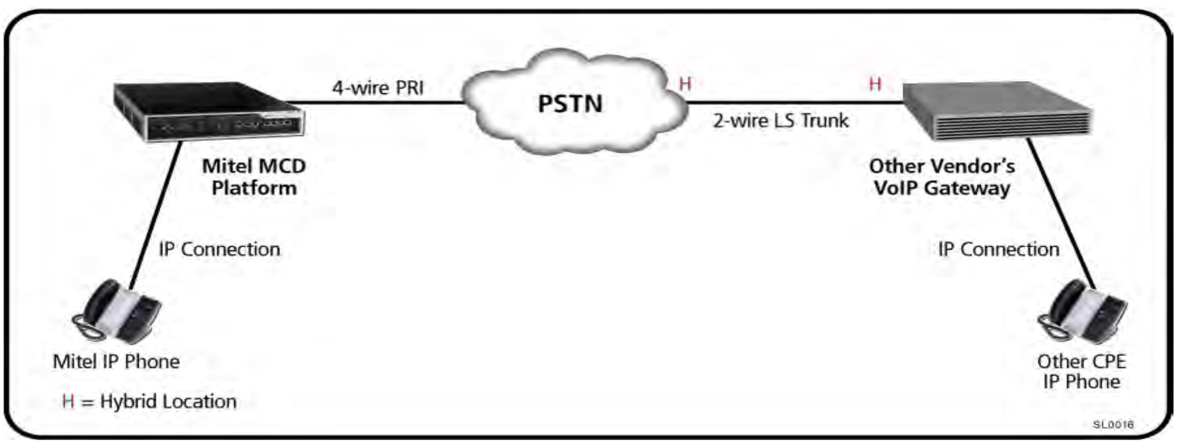

Figure 5: Scenario 3

In Scenario 3:

1. The PSTN is not aware of the equipment used on the MCD side of the PRI interface.
2. If the MCD system introduces an echo causing delay in the IP path, you are responsible for preventing (cancelling) the echo from entering the PSTN. Otherwise, echo will be heard by the CPE VoIP phone user. Even though the PRI interface is 4-wire, there could be a source of echo (for example, the LS trunk) on the other end of the connection.
3. There are two hybrids, one on each end of the 2-wire segment. This 2-wire segment contributes reflections towards the Mitel IP phone. The MCD echo canceller is responsible for cancelling reflections from sources that are bound for the Mitel IP phone. The VoIP

gateway is responsible for cancelling any echo delays that originate from the CPE VoIP phone.

4. These two hybrids also each generate a reflection in the direction of the other CPE VoIP phone. Since the delay through the VoIP gateway is long enough for these reflections to be heard as echo by the CPE VoIP phone, the VoIP gateway must cancel this echo.

5. If the PSTN introduces a delay that is long enough for the reflections to result in echo, then the PSTN is responsible for placing echo cancellers in the audio path. The PSTN will normally insert a pair of echo cancellers, one at each end. These echo cancellers will prevent 2-wire PSTN sets from hearing echo. The MCD also has echo cancellers behind the PSTN echo cancellers. If a PSTN echo canceller is operating properly, the MCD echo canceller does not detect any echo and it disables itself. If the PSTN echo canceller is faulty, the MCD echo canceller located downstream could make the echo worse.

6. There are no sources of electrical echo in the Mitel CPE, because all the internal paths are 4-wire.

7. Both the Mitel IP phone and the CPE VoIP phone could generate acoustic, mechanical, or crosstalk echo. The acoustic echo suppressors or acoustic echo cancellers on the handsets are responsible for cancelling this echo.

8. The equipment that inserts the echo causing delays is responsible for cancelling any reflections that would be heard as echo because of this delay.

Echo Cancellers and Echo Suppressors

The telecom industry has dealt with the problem of echo long before the introduction of VoIP technology. Satellite calls have echo because of the significant delay involved. Before Digital Signal Processing (DSP) technology, the way to eliminate echo was through the use of a device called an “echo suppressor.” Echo suppressors are essentially half-duplex switches that block audio in one direction. While echo suppressors do eliminate echo, they introduce the unpleasant side effect of a “switched” or half-duplex call.

Once DSP technology became readily available, echo cancellers became commonly used. Echo cancellers are simple in concept. They mirror the delay and the echo and then subtract them from the call. The key component in an echo canceller is the "adaptive filter” that adjusts itself to the signal error and determines the amount of delay and echo to cancel.

Cancel Echo Before it Enters the IP Cloud

Echo cancellers work on the assumption that echoes generally have

- a fixed delay path (that is, there is no variation in the delay)
- and a linear audio path (that is, the audio wave shape remains unchanged except for a constant loss).

In circuit-switched TDM telephony, communication paths are generally linear with fixed delay. Echo cancellers work well for cancelling echoes in the circuit-switched TDM domain.

In IP telephony, the communication path through the IP cloud is variable in delay (which is why there is a need for jitter buffers) and non-linear (because of packet loss and non-linear CODECs

such as G.729 and G.722.1). Echo cancellers cannot cancel echo effectively if the echo traverses an IP cloud as part of its round trip path.

This explains why echo cancellers must always cancel echo in the circuit-switched network before the echo enters the IP cloud. After the echo enters the IP cloud, the echo canceller cannot cancel echo effectively when it exits the IP cloud. See Figure 6 for an illustration.

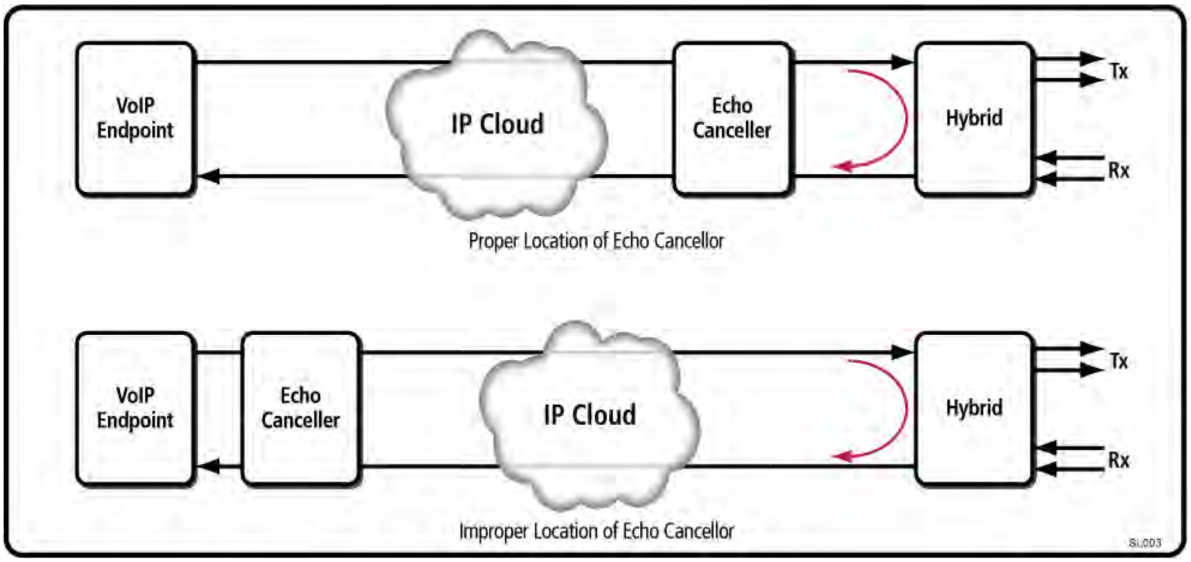

Figure 6: Echo Canceller Location

Echo Cancellers are Unidirectional

Another important point to note about echo cancellers is that they are unidirectional. They cancel echo in only one direction. In most scenarios, cancelling echo in one direction is all that is needed. However, in some special cases, it may be desirable to cancel reflections in both directions. In such a scenario, two echo cancellers would be needed, one for each direction.

Non-Linear Processing – NLP

Non-linear processing (NLP) is used by echo cancellers to suppress any residual echo that the adaptive filter does not cancel. As the name suggests, the NLP does not use linear processing (which the adaptive filter uses). The adaptive filter used in an echo canceller cannot usually fully cancel the echo, because there are usually some minor non-linearities in the echo path. Some low-level residual echo almost always remains. The NLP suppresses or blocks the remaining echo. There is a trade-off between using more or less aggressive NLP. When NLP is less aggressive, it may fail to remove echo if a large amount of echo remains after the adaptive echo cancellation filter. This issue may occur when there are large non-linearities, or when the adaptive filter does not accurately mirror the echo in the signal. When NLP is too aggressive, it can result in choppy audio if two parties are talking at the same time (double talk). The goal is to prevent both echo and choppy audio, but sometimes, the solution is to establish a balance between residual echo and choppiness.

Echo Cancellers in the Mitel Communications Platform

Echo cancellers are used in

- IP phones to cancel acoustic echo
- Line cards and analog option boards to cancel electrical (line) echo.

Causes of Perceived Echo

There are several conditions that can cause users to believe that they are hearing echo on a call.

Double Talk

Double talk occurs when both parties in a conversion talk at the same time. During double talk, the NLP (echo suppressor) is disabled. This is necessary to allow the other speaker to be heard. However, this also means that residual echo is not blocked. If the echo canceller filters are properly adapted, problems will not occur. However, double talk makes it difficult for the filters to maintain proper adaptation and sometimes it can cause the echo canceller to lose adaptation. In this case, the echo canceller quickly re-adapts when double talk is removed.

The type of double talk that occurs during a normal conversation should not degrade voice quality. However, it is possible to create echo, clipping, or a combination of the two, if users continuously talk over each other in an effort to stress the echo canceller. Echo that is artificially generated in this fashion should not be considered a voice quality issue.

Background Noise

There have been instances where a very loud background noise has caused people to believe that they are experiencing echo; for example, in rooms with noisy machinery, call centers with loud noise, or ringing. Essentially, this is another instance of double talk. Background noise can appear to the echo canceller as if someone were talking, thus causing the double-talk effect. This issue is more likely to occur on Bluetooth® headsets.

Cell Phones

We have identified instances when customers were hearing acoustic echo, usually during double-talk, while talking to people on a cell phone. This echo is due to a failure of the cell phone's acoustic echo canceller to cancel the echo. Because of the long delays and non-linear coding in the cellular network, only the echo canceller in the cell phone can cancel this echo.

Cell phone calls can also exhibit symptoms similar to echo when a change in cell phone towers occurs. A change in cell phone towers can cause a brief disturbance in the signal that sounds like echo which stops after a few seconds or less.

Two or More Echo Cancellers in Series

To cancel echo effectively, there should never be more than one echo canceller in each direction. However, you cannot always avoid two echo cancellers in series, because the PSTN will sometimes use echo cancellation, and there is no way to determine when this occurs.

In Figure 7, if echo canceller #2 is functioning correctly, echo canceller #1 will never have a chance to adapt and there will be little impact on audio quality. However, if echo canceller #2 sometimes allows echo to pass, the echo canceller#1 will try to adapt. This scenario could cause voice quality issues.

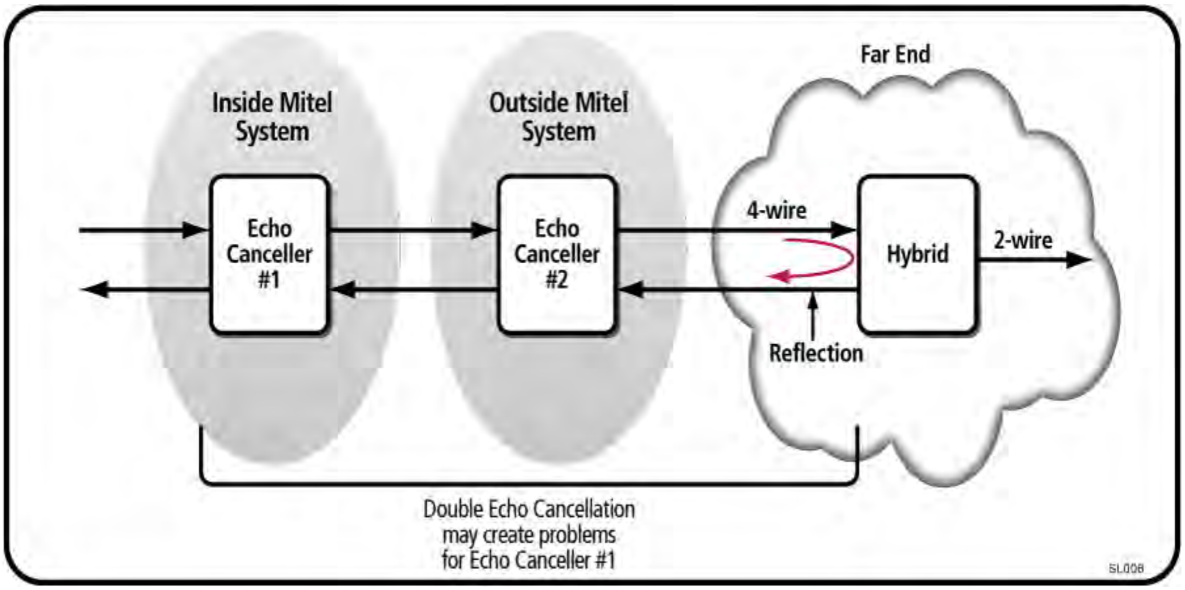

Figure 7: Double Echo Cancellation

Signal Level Mismatches

If the audio level from the far end is much lower than the transmitted audio level, the far end speech level may be close to the echo level. This causes the echo suppressor (NLP) to interpret the far-end speech as echo. As a result, the NLP may sometimes block low sections of the audio causing the incoming speech to sound choppy.

Cordless Headsets

Cordless headsets can have echo due to electromagnetic interference. Do not place the cordless headset's base station right next to the phone.

Scratchy Audio When Talking Loudly

We have experienced a few cases where people with a loud, deep voice believe that they are hearing echo. Echo cancellers work by analyzing linear signals to eliminate echo. If the audio signal becomes saturated, non-linearities can result, and the echo canceller will not be able to completely cancel the echo. This results in a scratchy, distorted signal when the person is

talking. This situation can happen if the speaker has a loud, deep voice that saturates the signal. Users will often report saturated audio as echo. See Figure 8 and Figure 9 for examples of normal and saturated audio.

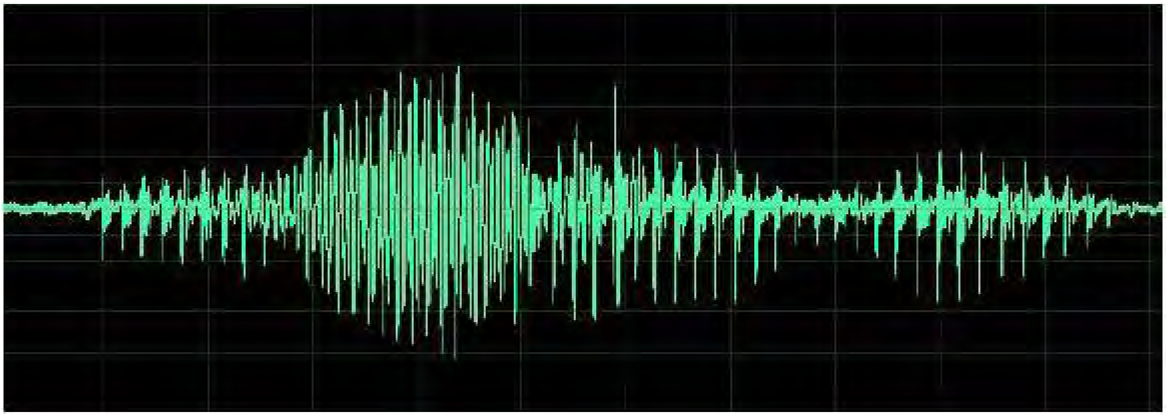

Figure 8: Normal Audio Sample

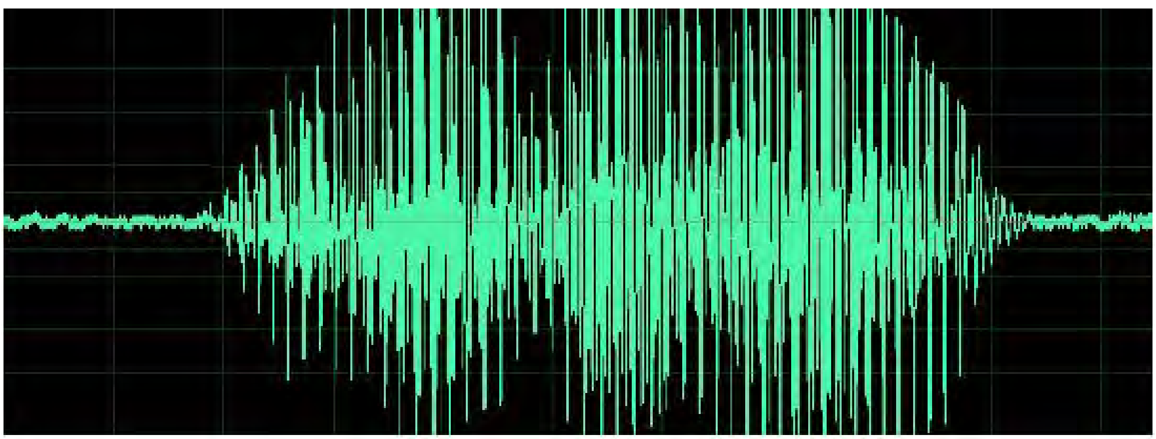

Figure 9: Saturated Audio Sample

Echo Basics Summary

Before you attempt to troubleshoot echo, you must understand the following points:

- The primary cause of echo is a combination of a significant delay plus a reflection. In order for echo to be heard by users, an echo canceller must fail to cancel or suppress the echo.
- Reflection causes echo if the round trip delay is long (greater that 50 ms round trip delay) and the reflections are not cancelled or suppressed. You must be able to identify the significant delays that are manifesting the reflections as echo. VoIP is a primary source of significant delay (amounting to several hundred milliseconds).
- Reflection sounds like sidetone when the round trip delay is short (for example, less than 50 ms).
- The two main causes of reflection are
  - 2-wire/4-wire interfaces, which are called hybrids, and

  - acoustic feedback from phones.
- You should be able to identify the types of echo:
  - Electrical (line) echo originates at hybrids which are always located at both ends of any 2-wire segment
  - Acoustic echo is caused by the failure of the echo canceller in the handset to suppress or cancel echo. It is more likely to occur when a phone, cell phone, or conference bridge is operating in handsfree mode.
- Conditions such as double talk, background noise, two or more echo cancellers in series, signal level mismatch, and overload distortion that occurs during loud audio can reduce the effectiveness of echo cancellers and can sometimes result in noticeable voice quality issues.
- Customers should be made aware of the following facts:
  - Reflections originating at Mitel IP phones that can cause echo must be cancelled by echo cancellers in the Mitel IP phone to prevent it from entering the long delay segment of the MCD VoIP audio path.
  - Reflections that originate in the short delay audio path of the PSTN must be prevented from entering the Mitel VoIP network by the echo cancellers in the Mitel MCD gateway.
  - Echo that is caused by reflection on the long delay segment of non-Mitel equipment on the other side of the PSTN cannot be cancelled by Mitel equipment. The owner of the non-Mitel equipment is responsible for cancelling any echo that originates at their phones from entering the PSTN network.

IP Network Issues

Conditions that can result in voice quality issues on IP Networks include:

- Latency (Delay)
- Packet Loss
- Excessive Speech Transcoding
- Jitter
- Lack of Bandwidth

Limits for Network Issues

Table 3 provides recommended network limits for end-to-end delay, ping delay, packet loss, and jitter.

Table 3: Network Limits

Rating	End-to-end delay	Ping delay	Packet loss	Jitter
Good	<50 ms	<100 ms	<0.5%	<20 ms
Borderline	<80 ms	<160 ms	<2%	<60 ms
Unacceptable	>80 ms	>160 ms	>2%	>60 ms

Latency (Delay)

Latency in Calls

Latency in a call is the amount of time it takes for the caller's voice to reach the other end of the connection. As latency increases in a conversation, it becomes increasingly difficult for users to sustain a normal two-way conversation. The conversation rapidly deteriorates from an interactive exchange to an "over-to-you" radio-style form of communication. Severe latency in the network can result in dropped calls.

Latency becomes noticeable in a call at a 80 ms to 200 ms delay and is radio-style at 400 ms delay. Assuming that jitter and packet loss are not an issue, end-to-end delay is the most likely cause of voice quality issues on VoIP calls. From International Telecommunication Union - Telecommunication Standardization Sector (ITU-T) recommendations, and based on practical experience, the end-to-end delay for a voice call should not exceed 80 ms.

Latency in the Data Network

Latency in the data network is amount of time it takes for a data packet to travel through the network. Latency in the data network results in latency in calls. It's important to note that latency in the network can change significantly depending on the

- condition of the network
- amount of available bandwidth
- path that is being used through the network
- amount of traffic

For example, latency may be high during the afternoon when network usage on the site peaks. During the period of peak network usage, the IP phone user may experience poor voice quality on calls. Later, when network usage subsides, latency isn't a problem and voice quality on calls is good again. Therefore, you need to perform ping tests when the voice quality issue is present to determine if latency is the cause.

You can use a ping test to determine if network latency is causing voice quality issues between two IP phones. See “Ping Test” on page 62 for instructions. ping delay is the time it takes for a data packet to travel from one IP address across the network to another IP address and back again (round-trip delay). A ping utility allows you to send a data packet from a PC to an IP address on the network. If the packet reaches the IP address it is sent back, and the utility displays the round-trip delay. Typically, the send and receive paths have equal delays. You can also check the **Delay** time for the IP phone in the **Voice Quality Statistics** form of the MCD System Administration Tool (see “Voice Quality Statistics for Mitel IP Phones” on page 63 for details) to determine if latency is an issue.

Jitter

Jitter is the variation in the packet delay. The major cause of jitter is network congestion. This occurs when the amount of data arriving at a node, including the source, destination, routers, and switches exceeds the capacity of the node to forward the data. In this case, data is stored in a buffer queue until the node is able to forward the data. The time that the data spends in this buffer before being forwarded is the major source of jitter.

While high quality network routers, switches, and Network Interface Cards (NICs) can generally forward packets at the network data rate; many routers, switches, and NICs are limited in the number packets they can forward per second. With short packets, such as VoIP packets, this can result in network congestion at data rates significantly lower than the network bus bandwidth.

High levels of jitter results in choppy voice quality on VoIP calls. To determine if jitter is the cause of voice quality issues:

- Measure jitter using a network analyzer, or
- Check the **Average Jitter** for the IP phone in the **Voice Quality Statistics** form of the MCD System Administration Tool (see “Voice Quality Statistics for Mitel IP Phones” on page 63 for details).

Packet Loss

When packet loss occurs during a VoIP call, tiny fragments of the conversation are lost. Although data networks can handle some packet loss, even a small amount of packet loss can have a noticeable effect on voice quality. Users may experience one or more of the following effects:

- Loss of words in a conversation
- Dropped calls (if packet loss is severe)
- Loss of intelligibility
- Fuzziness
- Choppy audio or clipping

Common causes of packet loss within the network include:

- Congestion at a connection that results in buffer overflow and data loss.
- Highly variable jitter that causes packets to arrive too late at a gateway or IP phone. The packets arrive too late to be used for voice and are therefore discarded.
- Out-of-sequence packets being discarded on WAN connections.
- Incorrect duplex settings on LAN connections that leads to data collisions and packet loss.
- Lack of network bandwidth.

A lack of network bandwidth on the LAN or WAN can delay voice packets from reaching their destination and result in packet loss. When a voice packet is delayed, the destination gateway loses real-time information. In this event, the destination gateway must presume the content of the missed packet. This presumption by the gateway causes the received voice to have different characteristics from the original transmitted voice. If a voice packet is delayed beyond the receiving gateway's ability to guess at the content, the gateway leaves a gap in the packet. This gap causes part of the transmitted speech to be lost at the receiving end, resulting in choppy voice.

Although some packet loss can be handled on an ongoing basis, bursts of packet loss will be noticed by users. A network with 0.1% packet loss over time sounds much different than a network with the same loss but occurring in bursts of three or more packets.

You can determine the **Packet Loss** at an IP phone in the **Voice Quality Statistics** form of the MCD System Administration Tool (see "Voice Quality Statistics for Mitel IP Phones" on page 63 for details).

Excessive Speech Transcoding

When you place a VoIP call, your analog voice signals are converted to digital data, compressed into packets, and then transported over the network. The conversion and

compression/decompression process is called "transcoding" and is performed by coder/decoders (CODECs).

Note: The terms "transcoding" and "compression" are often used interchangeably, but this usage is incorrect. Transcoding is the direct digital-to-digital conversion of one voice digitization standard to another. It occurs when voice information is changed from one CODEC type to another. Compression simply refers to a reduction in the amount of data.

The amount of voice compression applied by the CODEC determines the quality of the voice signals. Voice compression works by removing a certain amount of information that is deemed non-essential (for example, not sending data during silent periods and sending only the main voice frequency elements rather than the full bandwidth). As a result, some voice information is lost. The quality of compressed voice is never as good as uncompressed voice, but intelligibility will typically be maintained.

There are several commonly used ITU-T standards used by CODECs for voice digitization:

- **G.711** is the default standard for Mitel communication platforms and for the PSTN. This standard digitizes voice into a 64 Kbps data stream. The G.711 standard does not compress voice. In a LAN environment where bandwidth is plentiful, G.711 should be used.
- **G.729a** is also supported by Mitel communication platforms for compressed voice operating at 8 Kbps, with 8 to 1 compression. With voice quality just below that of G.711, it is the second most commonly implemented standard. In a WAN environment, where access bandwidth is limited, G.729a can be used to increase the amount of voice traffic supported on a particular link. However, the recommended approach is to use G.711 with higher voice quality and a lower amount of voice traffic.
- **G.722.1** operates at 64 Kbps, but offers high-fidelity speech. Whereas the two previously described standards deliver an analog sound range of 3.4 kHz, G.722 delivers 7 kHz. This standard is supported on MCD Release 5.0 and later platforms.

Compression affects a number of call connection types. These include:

- IP phone to IP phone
- IP phone to TDM and vice versa
- IP phone at a remote site back to TDM or IP
- IP connection across an IP trunk route

Compression affects other aspects of the MCD system as well. These include IP phones, the MCD platform, MCD devices, IP applications, IP networking routes and trunk routes, and licenses. Voice subjected to excessive compression or too many transcoding hops (for example, from G.711 standard to G.729.a and back to G.711) sounds like the voice is coming through a drain pipe.

Lack of Network Bandwidth

Poor voice quality can result from a lack of network bandwidth. Ensure that there is sufficient bandwidth on the LAN and on the links to the WAN to support the amount of expected traffic. Refer to the *MCD Engineering Guidelines* for the LAN and WAN bandwidth requirements.

Environmental Issues

Issues in the user's environment or in the equipment environment can result in voice quality degradation. Environmental issues can occur in the analog, TDM, PSTN, or IP network.

Background Noise

Background noise in a caller's environment can cause the party at the other end to assume that they are hearing echo. Loud machinery, noisy crowds, a vibrating fan, or wind noise are all examples of background noise that can sound like echo to the caller on the other end.

Background noise on a call normally varies in strength as the noise level of the source fluctuates. This issue is associated with a specific destination and isn't present when you call other numbers.

Note: Wideband CODEC calls are more likely to be affected by background noise than G.711 or G.729 calls.

Voltage Interference

Voltage interference on analog or VoIP calls can cause buzzing.

Buzzing on Analog Calls

This issue is usually caused by poor grounding on the controller. The buzzing heard by users is typically constant.

Buzzing on VoIP Calls

This issue is usually caused by interference from power cables that have been routed alongside the Ethernet cables that connect the MCD controller or layer 2 switch to the IP phone. This type of voltage interference affects the Ethernet cable run from the switch to the IP phone because the IP phone uses a floating ground. This problem does not occur on Ethernet cables between switches and routers because they use earth grounds.

Note that the buzzing can be constant or intermittent depending on the whether the power cables are constantly energized (power feed to a server farm) or are energized intermittently (for example, whenever an air conditioning unit is running).

Configuration Errors

This section covers common configuration errors in network equipment that can affect voice quality. Configuration errors can be present in the following:

- MCD platforms and gateways
- Layer 2 Ethernet switches and routers

Configuration Errors on the MCD Platform and Gateways

The following configuration errors on the MCD platform and trunking gateways can affect voice quality:

- **Loop Start Trunk Settings Incorrect**: Refer to “Line Measurement Tool” on page 56 for instructions on how to correct this issue.
- **Wrong Loss and Level Plan in Use**: Check the **Country** setting in the **License and Options Selection** form in the MCD System Administration Tool. The **Country** setting determines the default language, dialing plan, tone plan, and loss and level plan. Refer to the *MCD Hardware Technical Reference Guide* for loss and level plan details.
- **Audio Level Settings Too High or Too Low**: Ensure that ONS/OPS Station Descriptor parameters in the MCD System Administration Tool are set correctly. If this issue is occurring at a specific phone, ensure that the volume on the phone is set to a normal level using the volume up/down keys.
- **Bandwidth Management on MCD Platform Programmed Incorrectly**: Refer to "Bandwidth Management" in the *System Administration Tool* online help for programming instructions.

Configuration Errors on the Layer 2/3 Switches and Routers

The MCD platform layer 2 Switch, and the network layer 2 switches, layer 3 switches, and routers must be configured correctly to avoid voice quality issues. Follow the guidelines in this section.

General Guidelines

- ❒ The LAN should be fully switched.
- ❒ Ensure that IP phones are connected to the IP network only through layer 2 switches. A hub works by sharing bandwidth among a number of devices. The devices use CSMA/CD to control access, but effectively “fight” each other for access. If you use hubs to connect IP phones to the network, there is a high probability of introducing unnecessary delay and jitter into VoIP calls.
- ❒ The MCD platform Ethernet ports default to auto-negotiate. Ensure that the Network Layer 2 switch ports are also configured to auto-negotiate.
- ❒ Use full-duplex connections wherever possible.

- ❐ Use VLANs to segregate voice packets from data traffic.
- ❐ Make proper use of Layer 2 and Layer 3 QoS mechanisms for wired LANs; make proper use of WMM QoS mechanisms for WiFi LANs.
- ❐ Provision extra bandwidth whenever possible.

VLANs

- ❐ With dual-port phones, it is important to configure the LAN switches to use tagging for the voice VLAN and no tagging for the default VLAN. This ensures that voice packets are properly prioritized over data applications from the PC.
- ❐ The IP phone Ethernet ports are hard-coded to auto-negotiate. Ensure that the network layer 2 ports are also configured to auto-negotiate.
- ❐ Configure ports to provide VLAN tagging to incoming untagged information and remove this tagging when passing out of the switch. This is used by the controller and associated applications.
- ❐ Configure ports to pass all active VLANs with tagging from one switch to another (there is no untagged information present in the connection). This maintains priority information between LAN switches.
- ❐ Configure ports to
  - accept tagged information, and
  - accept untagged information and pass it on to a specified VLAN.
- ❐ Do not use VLAN 0. The specification standard for VLAN 0 leaves room for interpretation. This can lead to incompatibility between different vendor units.
- ❐ Set Priority to value 6 for voice (5 for Cisco switches).
- ❐ Set Priority for untagged VLAN/native VLAN/default_vlan to 0.
- ❐ Hubs do not support priority queuing, so use managed layer 2 switches with 802.1p/Q support.
- ❐ Do not use VLAN 1000 and above with Cisco products.
- ❐ Do not use VLAN 4095 with HP products; this is reserved for inter-switch use.
- ❐ Do not use VLAN 4094 with the CXi controller.
- ❐ Do not share the voice VLAN with data devices.

WAN Layer 3 Switches

- ❐ Ensure that the packet per second (PPS) rate of routers and switches is adequate for the amount of voice traffic.
- ❐ Set the router Maximum Transmission Units (MTU) value to divide larger packets into smaller packets, especially for slower speed links (anything less than T1 rates). See the *MCD Engineering Guidelines* for details.

LAN Quality of Service Policies

The MCD platform can apply different LAN QoS policies to voice packets, signaling packets and other packets. Use the **LAN Policy (QoS)** form in the MCD System Administration Tool to set the LAN QoS policy values. In a Cisco-based environment the recommended settings are:

- Voice Packets: DSCP: 46, 802.1p:5
- Signaling Packets: DSCP: 26, 802.1p:3
- Other Packets: DSCP:0 802.1p:0

Subnets

- Subnet your network to reduce broadcast traffic.
- Include layer 3 devices to improve speed within communities of interest and the overall-network, and reduce the burden on the system to all broadcast traffic. It is also a requirement for VLANs to operate correctly and provide the voice priority required when using dual-port phones.

Firewall Settings

Ensure that the ports on the MCD platform are configured in the System Administration Tool Port Forward Table to allow external traffic to reach resources on the internal network. The table can contain up to 40 entries. Refer to the *MCD Engineering Guidelines* for firewall details and a complete list of the MCD ports.

Table 4: Port Forward Table (CXi/CXi II/MXe/MXe Server only)

Parameter	Function/Values
Protocol	The WAN interface protocol; UDP or TCP.
Src Start Port	The source port at the start of the range.
Src End Port	The source port at the end of the range.
Dst IP Address	IP Address of the destination device.
Dst Start Port	Destination port at the start of the range.
Dst End Port	Destination port at the end of the range.

Troubleshooting Voice Quality

Methodology

When troubleshooting voice quality issues, follow the steps listed below:

1. Record site information (contact details, hardware and software, ARS information).
2. Collect problem details and list affected devices.
3. Create a detailed network diagram.
4. Isolate the voice quality issue.
5. Use the troubleshooting charts to determine the corrective action.

Record Site Information

Contact Details

Table 5: Contact Details

Details		Record Information
Company	Name	
	Address	
	State/Province/Country	
	ZIP/Postal Code	
	Time zone	
Site Contact	Name	
	Telephone	
	Cell Phone	
	Email Address	
Authorized Partner information	VAR/Partner name	
	Address	
	City	
	State/Prov/Country	
	ZIP/Postal Code	
	Time Zone	

Hardware and Software Information

Use the following table to record the

- Site hardware platforms (for example, MCD platforms, MCD gateways, Analog Service Units, peripheral nodes)
- Software versions (for example, MCD Release 4.2) running on each platform.
- PSTN Connectivity (LS, PRI, SIP, and so forth)
- Loss and Level Plan in use (for example, North America or United Kingdom)

Table 6: Record Hardware and Software Information

Platform	Software Version	PSTN Connectivity	Loss Plan

Collect Problem Details

Speak to the users on the customer site and record the problem symptoms. If you are on site, place some calls and experience the voice quality issues first hand. If possible, use a handset audio tap tool to obtain a recording of the call. Table 7 provides a list of basic questions.

Note: It's important that you collect the problem details without making any immediate assumptions. Also, user input, although necessary, can also be misleading. Although a user may complain that all their calls have voice quality issues, the issue may in fact be isolated to a certain type of call (for example, VoIP to VoIP calls only) or to a particular destination.

Note: When completing this table, check all boxes that apply. However, do not check a box unless you are sure that the problem is present.

Table 7: Collecting Problem Details

Voice Quality Questions	Problem Details
What is the nature of the problem? **Tip**: You can find sample audio recordings of some of these voice quality issues on the internet at the following URL: http://www.voiptroubleshooter.com/diagnosis/usersymptoms.html	❒ Buzzing
	❒ Choppy voice
	❒ Clipped voice (truncated words)
	❒ Clipped voice (overdriven or distorted)
	❒ Crackling or clicking
	❒ Crosstalk
	❒ Delayed response time (effect similar to Citizen’s Band radio)
	❒ Distortion
	❒ Distorted music on hold
	❒ Dropped calls
	❒ Echo
	❒ Garbled voice
	❒ Hissing
	❒ Hum
	❒ Motor boating
	❒ Popping - during and after speech
	❒ Popping - random
	❒ Sidetone (too little)
	❒ Silence (no audio on both ends)
	❒ Silence (one-way audio)
	❒ Static
	❒ Stuttering
	❒ Tinny audio (as though voice is coming through a drainpipe)
	❒ Poor intelligibility
	❒ Other, specify

Table 7: Collecting Problem Details

Voice Quality Questions	Problem Details
Is it loud or soft?	❒ Loud ❒ Soft ❒ Varies from loud to soft ❒ None (no volume issues)
Is the problem constant or intermittent?	❒ Constant ❒ Intermittent during some calls ❒ Intermittent during all calls. ❒ Intermittent during the day ❒ Intermittent during the week ❒ Specific time of day
When did the problem first appear?	Date: Time:
Where is the problem occurring?	❒ Local only ❒ Far-end only ❒ Both locally and far-end ❒ Don't know
What is the customer environment?	❒ Enclosed Office ❒ Open concept office ❒ Tenanting (multiple companies in same building) ❒ Warehouse ❒ Manufacturing ❒ Academic institution (for example, college campus) ❒ Hospital ❒ Hospitality (hotel or motel) ❒ Call center (ACD environment) ❒ Construction site ❒ Out of doors

Table 7: Collecting Problem Details

Voice Quality Questions	Problem Details

What type of calls are affected?	❒ Internal only
	❒ External only
	    ❒ Inbound only
	    ❒ Outbound only
	    ❒ Both
	❒ Internal and External
	❒ Handsfree
	❒ Wireless
	    ❒ Mobile Voice System (MVS)
	    ❒ Digital Enhanced Cordless Telecommunications (DECT)
	    ❒ IP DECT
	❒ Conference
	❒ Teleworker
	❒ Mitel Collaboration Advanced (MCA)
	❒ Unified Communicator Advanced (UCA)
	❒ All types

Is there a specific type of call destination associated with the problem?	❒ Local ❒ Long-distance ❒ Cell phone ❒ Another PBX vendor
How many phones are affected?	❒ One ❒ Group of phones ❒ All phones
Is there a specific trunk exhibiting the problem?	SMDR Trunk ID ______
What type of trunk?	❒ Analog ❒ Digital ❒ Mitel IP ❒ SIP
How long has system been in service?	______________________________
Have there been any recent changes?	❒ Software upgrades ❒ Recent moves, adds, or changes of users ❒ New telephony equipment ❒ Other new equipment ❒ Changes in provider services ❒ LAN or networking changes Provide details of changes: ____ ____

Use the following table to list the devices affected.

Table 8: Affected Devices

Device Type	Extension	Call Type	Trunk ID	Symptom	Date/Time	Local or Long Distance	Volume (loud or soft)
5330 IP Phone	5001	External Only	512	Constant Buzzing	May 5, 2011 10:00 AM	Long Distance	Loud

Create a Detailed Network Diagram

A network diagram will help you to pinpoint potential sources of voice quality issues. For example, electrical echo originates at the ends of 2-wire segments.

Create a diagram that shows both the analog and VoIP network components. When you create the diagram, ensure that you show the following:

- Communication Platforms: Identify the number and type of communication platforms (for example: vMCD, MICD, MCD on ISS, MXe, CX/CXi, AX controllers). If the system is resilient, identify the primary and secondary controllers.
- Analog Service Units (ASUs) and controllers with main option boards and embedded analog option boards
- Network Service Units
- Management platforms (for example, Mitel Applications Suite or Enterprise Manager)
- Phones: IP, Digital, Analog, Wireless
- ONS Lines
- PCs: Dual Port IP phones with connected PCs
- Analog Trunks: Label the trunks as (1) Digital; (2) 2-wire audio analog; (4) 4-wire audio analog
- IP Trunks
- SIP Trunks
- VoIP Trunking Gateways: Identify the gateway type
- Layer 2 Switches
- Routers
- Firewalls
- IP Addresses
- Subnets
- VLANs
- Bandwidth on data links that connect network components
- Bandwidth Management Zones
- Quality of Service Levels

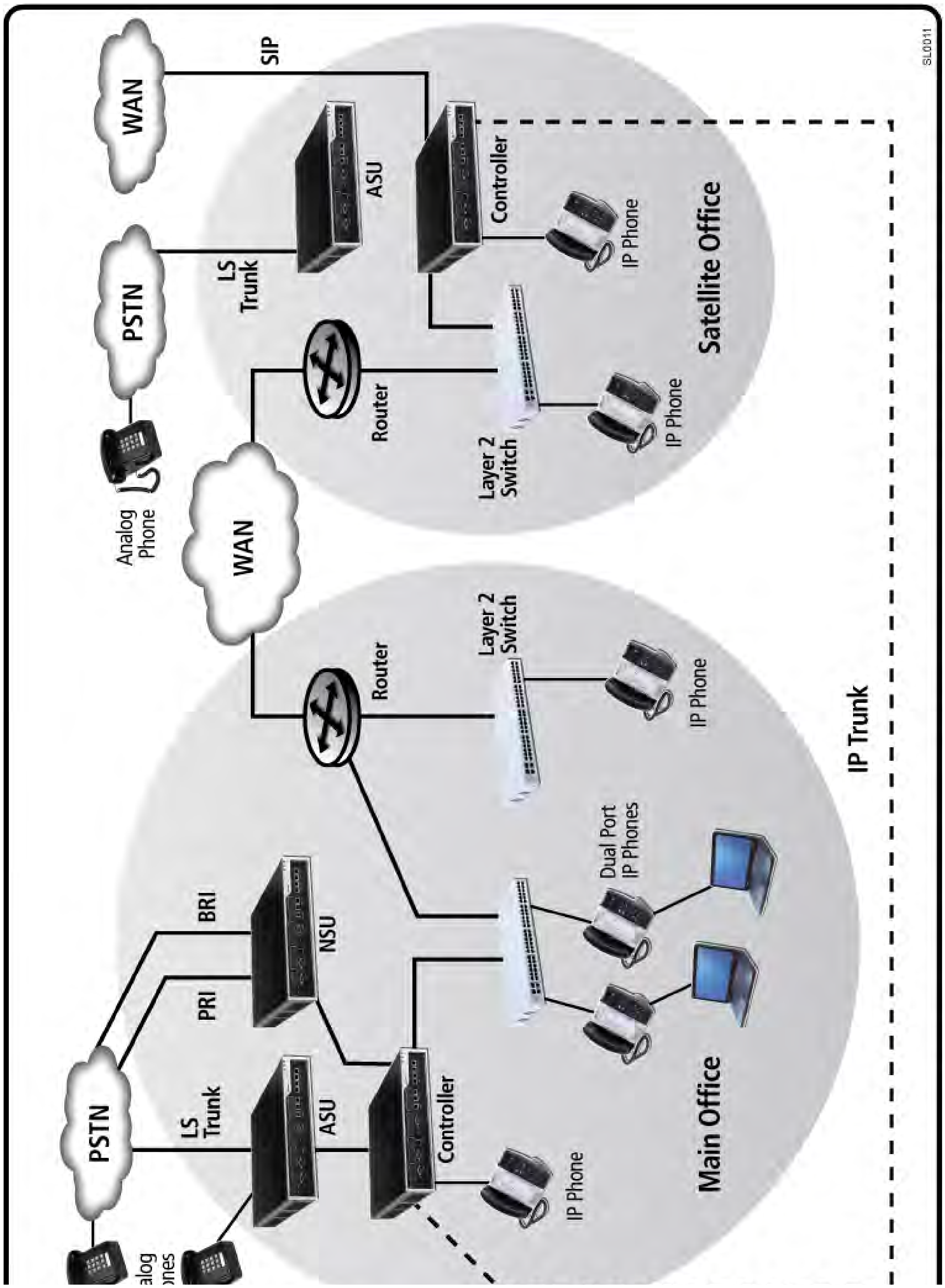

Figure 10: Example of a Simple Network Diagram

Isolate the Issue

Isolate the voice quality issue to a specific component (for example, IP phone, analog phone, controller, switch, or router) or to an area of the network (for example, ONS line, analog trunks, WAN, LAN, subnet, or vLAN) by placing calls to destinations that use different paths through the network. If the issue is constant and the call path does not exhibit the voice quality issue, then you can exclude the call path from your troubleshooting efforts. However, if the issue occurs intermittently, you will need to perform multiple tests on the call path in order to rule it out. The goal is to narrow the scope of your troubleshooting to the parts of the network where the issue is occurring.

Figure 10 provides an example of this troubleshooting methodology. By placing calls to different destinations, you determine that the voice quality issue is not present on calls

- between IP phones on the LAN at the main office
- between IP phones on the LAN at the satellite office
- between analog phones on the ASU or PSTN
- over IP, BRI, or PRI trunks.

In this example, you determine that the voice quality issue occurs only on VoIP calls between IP phones located in the main office and those in the satellite office. In this case, you would focus your troubleshooting on the

- routers at the network edges
- available bandwidth and traffic occurring on the path
- controller in the satellite office.

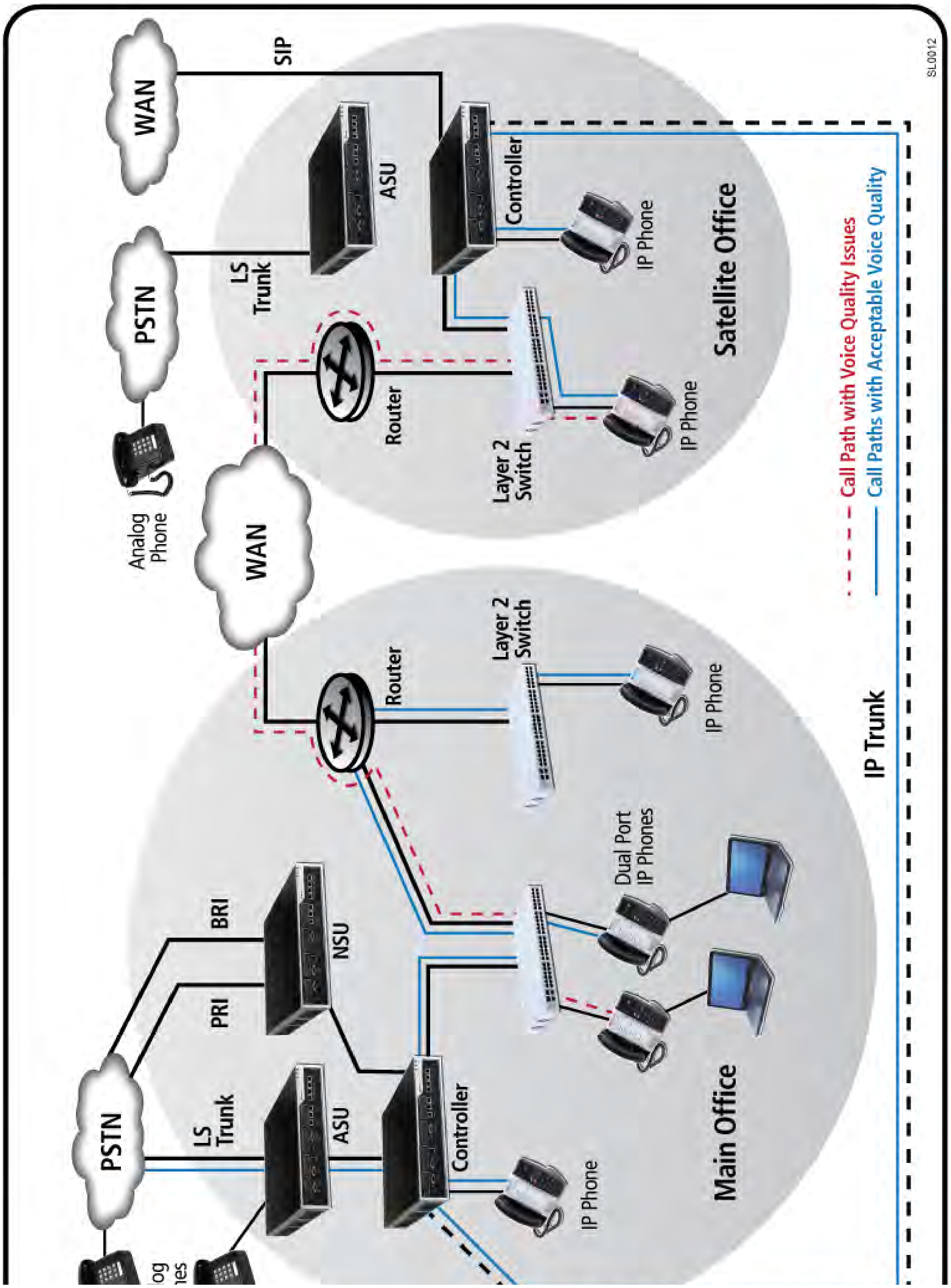

Figure 11: Example of Isolating Voice Quality Issue

After isolating the issue, use Table 9 to record where the voice quality issue is occurring in the network. Check all that apply.

Table 9: Isolating the Voice Quality Issue

The issue is occurring on . . . .
❒ Single IP Phone? ❒ on a Dual port IP phone that connects a PC to the network
❒ Multiple IP Phones? ❒ on the same MCD platform? ❒ in the same subnet ❒ between IP phones on different subnets ❒ on the same layer 2 switch ❒ in the same VLAN ❒ on the same router ❒ in the same MCD bandwidth management zone
❒ Single Analog or Digital Phone?
❒ Multiple Analog or Digital Phones? ❒ on the same ASU ❒ on the same MCD platform?
❒ IP Phone to Analog/Digital Phone Calls or Vice-Versa? ❒ across the PSTN ❒ between analog phones on ASU and IP phones in LAN
❒ IP Trunk Calls? ❒ on a single IP trunk ❒ on calls on the same MCD platform ❒ on calls across a specific MCD trunking gateway ❒ between the main and satellite office ❒ across the WAN
❒ SIP Trunk Calls
❒ Analog or Digital Trunk Calls? ❒ on all trunks on an ASU ❒ on all trunks on a specific trunk card ❒ on all trunks in a specific trunk group

Now that you have . . . .

- ❒ recorded the site information
- ❒ collected problem details
- ❒ identified the symptoms, and
- ❒ isolated the voice quality issue

. . . . you are ready to proceed to the troubleshooting charts.

Troubleshooting Charts

Based on the problem symptoms, see the corresponding chart:

- “Echo” on page 42
- “Hissing, Buzzing, Hum, Motor Boating” on page 44
- “Crackling, Clicking, Popping, or Static” on page 45
- “Choppy Voice” on page 47
- “Clipping (Truncating during Double-Talk)” on page 48
- “Stutter” on page 48
- “Garbled Voice, Poor Intelligibility, or Distortion” on page 49
- “Crosstalk” on page 50
- “Dropped Calls” on page 51
- “Silence” on page 51
- “Tinny Audio (Drain Pipe)” on page 52
- “Delayed Response Time (Citizen’s Band Effect)” on page 53
- “Low or High Audio” on page 54

Echo

Note that echo can be

- Constant
- Intermittent
- Residual (very low level or quiet echo)
- Distorted buzz that sounds like echo (the signal is saturated resulting in distorted audio and the echo canceller is unable to remove the distortion).

Table 10: Electrical Echo

The issue is occurring on . . .	Probable Cause	Corrective Action
Multiple IP phones on calls across LS trunks	Impedance mismatch	Run the Line Measurement Tool (see page 56) to apply correct settings.
IP Phone to IP phone calls, or IP phone calls across TDM.	Packet loss or bit errors occurring due to incorrect Quality of Service settings.	Ensure Quality of Service setting are correct on all network components (MCD platforms, layer 2 switches, layer 3 switches, routers, and VLANs). Refer to the *MCD Engineering Guidelines* for QoS guidelines.

Table 10: Electrical Echo

The issue is occurring on . . .	Probable Cause	Corrective Action
IP phone to IP phone calls and IP phone calls across TDM	Excessive latency causing sidetone to be perceived as echo. The excessive latency is most likely occurring in the VoIP network, either in the Mitel IP system, or on the CPE VoIP system on other side of the PSTN.	Add bandwidth.
		Configure Bandwidth Management zones. Refer to the *MCD Engineering Guidelines* and System Administration Tool online help for details.
Multiple analog and IP phones	Faulty echo canceller in an Analog Services Unit (ASU), on an MCD Analog Main Board (AMB), or on an MCD Analog Options Board (AOB).	Replace ASU, AMB, or AOB.
Multiple analog and IP phones when calling to or being called from a specific external destination.	Echo is originating at CPE on other side of PSTN.	Refer to “Identifying the Source of Echo” on page 55 to confirm that the echo is originating in the far-end CPE. Inform customer that this issue is not being caused by Mitel equipment

Note: If you hear acoustic echo on a call, it is originating at the far-end phone.

Table 11: Acoustic Echo

The issue is occurring on . . .	Probable Cause	Corrective Action
Single phone in handsfree mode (intermittent)	Caller is using phone in handsfree mode or caller is using a speaker phone.	Reduce volume or switch to handset mode.
	Environment is too reflective.	Reposition phone away from walls, cabinets, shelves (any hard surfaces that may reflect sound).
	User has placed the handset down on the desk facing the set microphone. Audio from the handset speaker is entering the set microphone.	Ensure that the handset is not placed near the set microphone when operating in handsfree mode.
Single phone in handset mode (constant)	Sidetone is too loud and is being perceived as echo by user.	Reduce volume.
	Faulty handset (potentially an issue with Rev C handsets)	Test with a known working handset. If necessary, replace faulty handset.
	Faulty phone	Test with known working phone. If necessary, replace phone.

Table 11: Acoustic Echo

The issue is occurring on . . .	Probable Cause	Corrective Action
Multiple analog and IP phones when calling a specific external destination or remote office.	Echo is originating at CPE on other side of PSTN.	Confirm that the problem originates at the far end. See “Identifying the Source of Echo” on page 55 for instructions. Inform customer that this issue is not being caused by Mitel equipment.
Multiple analog and IP phones when calling a cell phone, or device that uses Bluetooth technology..	Echo is originating at cell phone or Bluetooth device.	

Hissing, Buzzing, Hum, Motor Boating

Table 12: Hissing, Buzzing, Hum, Motor Boating

The issue is occurring . . .	Probable Cause	Corrective Action
All external calls on analog phones	Electrical noise caused by grounding issue in wiring closet where PSTN trunks connect to main distribution frame.	Ensure that main distribution frame is properly connected to ground. Refer to "Cabling Guidelines in the Equipment Room" in Appendix A in the *MCD Hardware Technical Reference Manual* for guidelines.
All calls involving analog phones on a specific ASU or MCD controller.	Electrical noise caused by grounding issue on ASU or MCD	Ensure that the ASU or MCD controller is properly connected to ground. Refer to "Cabling Guidelines in the Equipment Room" in Appendix A in the *MCD Hardware Technical Reference Manual* for guidelines.
On multiple analog phones connected to a secondary main distribution frame	Electrical noise caused by grounding issue at a secondary main distribution frame	Check the ground at secondary main distribution frame.
On multiple IP and analog phones in a common location within a building (for example, a specific floor or area)	Electrical interference caused by routing the Ethernet cables or phone lines close to power cables (for example, routing cables horizontally in the same cable channel or vertically together in an elevator shaft).	Route power cables separately from phone lines and Ethernet cables. Refer to "Ethernet Cabling Guidelines in Appendix A of the *MCD Hardware Technical Reference Manual*.
On multiple phones across network.	Low volume hissing is typically channel noise caused by signal energy (white noise).	None. All calls have some idle channel noise.

The issue is occurring . . .	Probable Cause	Corrective Action
All calls involving a single IP phone	Ethernet cable that connects IP Phone to MCD controller or layer 2 switch is routed in close proximity to power conductors. Electrical interference from power cable is resulting in a voice quality issue.	Route Ethernet cables separately away from any power cables. Refer to the *MCD Hardware Technical Manual* for Ethernet cabling guidelines.
	Ethernet cable that connects phone to network connector is too long.	Ensure the cables used for IP phone connections are a maximum of 1.5 m (5 ft) in length. Refer to the *MCD Hardware Technical Manual* for Ethernet cabling guidelines.
On all calls involving one or more IP phones	Hum is occurring on calls due to faulty power over Ethernet (PoE) from layer 2 switch.	Check PoE settings on layer 2 switch. Install a power brick on the phones that are experiencing the humming issue.
	Hum is occurring on call because the power brick is faulty.	Use PoE from layer 2 switch or replace power brick.

Crackling, Clicking, Popping, or Static

Table 13: Cracking, Clicking, Popping, or Static

The issue is occurring on . . .	Probable Cause	Corrective Action
Calls involving IP phones on LAN or WAN.	Packet loss due to lack of network bandwidth between network components (MCD platforms, layer 2 switches, layer 3 switches, or routers).	Configure Bandwidth Management zones. Refer to the *MCD Engineering Guidelines* and System Administration Tool online help for details.
		Add bandwidth.
	Packet loss or bit errors occurring due to incorrect Quality of Service settings.	Ensure Quality of Service settings are correct on all network components (MCD platforms, layer 2 switches, layer 3 switches, routers, and VLANs). Refer to the *MCD Engineering Guidelines* for QoS guidelines.
	Unsupported CODECs	Verify Service Level Agreements with WAN link vendor and ensure that the vendor supports the CODECs. Also ensure that they correctly match the priority bits that you are using.
	Excessive collisions caused by auto negotiate mismatch	Check switch settings.
	Excessive collisions caused by insufficient cabling. For example, 10000 Mbps being negotiated across a CAT 5 cable that can only really support 100 Mbps.	Upgrade cabling.

Table 13: Cracking, Clicking, Popping, or Static

The issue is occurring on . . .	Probable Cause	Corrective Action
Calls involving IP phones on a specific MCD platform, layer 2 switch, layer 3 switch, or router.	Packet loss or bit errors occurring due to incorrect Quality of Service settings on MCD platform, layer 2 switch, layer 3 switch, or router.	Ensure Quality of Service settings are correct on all network components (MCD platforms, layer 2 switches, layer 3 switches, routers, and VLANs). Refer to the *MCD Engineering Guidelines* for QoS guidelines.
Calls involving IP phones that are connected to a hub.	Packet loss occurring due to contention between the devices on the hub.	Mitel IP Phones are not supported on hubs. Replace hub with layer 2 or layer 3 switch.
On multiple IP and analog phones in a common location within a building (for example, a specific floor or area).	Electrical interference caused by routing the Ethernet cables or phone lines close to power cables (for example, routing cables horizontally in the same cable channel or vertically together in an elevator shaft).	Route power cables separately from phone lines and Ethernet cables. Refer to the *MCD Hardware Technical Manual* for Ethernet cabling guidelines.
On multiple phones across network.	Low volume hissing is typically channel noise caused by signal energy (white noise).	None. All calls have some idle channel noise.
All calls involving a single IP phone	Ethernet cable that connects the IP Phone to the MCD controller or layer 2 switch is routed in close proximity to power conductors. Electrical interference from power cable is resulting in a voice quality issue.	Route Ethernet cables separately, away from any power cables. Refer to the *MCD Hardware Technical Manual* for Ethernet cabling guidelines.
	Ethernet cable that connects the phone to the network connector is too long.	Ensure that the cables used for IP phone connections are a maximum of 1.5 m (5 ft) in length. Refer to the *MCD Hardware Technical Manual* for Ethernet cabling guidelines.
IP or TDM calls over PRI links.	Faulty cabling or faulty programming.	Refer to "Digital Trunk Troubleshooting" in the *MCD Troubleshooting Guide*.

Choppy Voice

Table 14: Choppy Voice

The issue is occurring on . . .	Probable Cause	Corrective Action
Calls involving multiple IP phones on LAN or WAN.	Excessive latency, excessive jitter, or packet loss due to lack of network bandwidth between network components (MCD platforms, layer 2 switches, layer 3 switches, or routers).	Add bandwidth.
		Configure Bandwidth Management zones. Refer to the *MCD Engineering Guidelines* and System Administration Tool online help for details.
	Packet loss or bit errors occurring due to incorrect Quality of Service settings.	Ensure Quality of Service settings are correct on all network components (MCD platforms, layer 2 switches, layer 3 switches, routers, and VLANs). Refer to the *MCD Engineering Guidelines* for QoS guidelines.
Calls involving IP phones on a specific MCD platform , layer 2 switch, layer 3 switch, or router.	Packet loss or bit errors occurring due to incorrect Quality of Service settings on MCD platform , layer 2 switch, layer 3 switch, or router.	Ensure Quality of Service settings are correct on all network components (MCD platforms, layer 2 switches, layer 3 switches, routers, and VLANs). Refer to the *MCD Engineering Guidelines* for QoS guidelines.
Calls involving wireless (Bluetooth, DECT, or Wi-Fi) phones or cell phones.	Strength of wireless signal fades as user moves around site.	Ensure wireless zone coverage is adequate using the wireless site survey. Add base stations if required.
		Instruct users to stay within range of base stations.

Note: Use the following tools to evaluate bandwidth on the network:

- “Network Traffic Analyzer” on page 59
- “Voice Quality Statistics for Mitel IP Phones” on page 63
- “VoIP Bandwidth Management on MCD Platforms” on page 60
- “Ping Test” on page 62

Clipping (Truncating during Double-Talk)

Table 15: Clipping

The issue is occurring on . . .	Probable Cause	Corrective Action
Multiple IP phones on calls across LS trunks.	Excessive echo suppression in the network.	Run the Line Measurement Tool (see page 56) to apply correct settings.
Phone in handsfree mode.	Speakerphone volume is too high.	Reduce volume of speakerphone.
Calls involving a specific user.	User is speaking into the phone using a particularly loud and low voice.	Use a Network Traffic Analyzer, such as Wireshark, to capture a sample of the audio at both ends of the call. Compare the samples. If the sample from one caller appears saturated (see Figure 9 for an example), instruct the caller to speak in a normal tone of voice.

Stutter

Table 16: Stutter

The issue is occurring on . . .	Probable Cause	Corrective Action
Calls involving IP phones on a specific MCD platform, layer 2 switch, layer 3 switch, or router.	Quality of Service settings are incorrect, resulting in a packet bottleneck.	Ensure Quality of Service settings are correct on the specific network component (MCD platforms, layer 2 switches, layer 3 switches, routers, and VLANs). Refer to the *MCD Engineering Guidelines* for QoS guidelines.
Calls involving multiple IP phones across LAN or WAN.		Ensure Quality of Service settings are correct on all network components (MCD platforms, layer 2 switches, layer 3 switches, routers, and VLANs). Refer to the *MCD Engineering Guidelines* for QoS guidelines.

Garbled Voice, Poor Intelligibility, or Distortion

Table 17: Garbled Voice, Poor Intelligibility, or Distortion

The issue is occurring on . . .	Probable Cause	Corrective Action
Calls involving IP phones on LAN or WAN.	Packet loss due to lack of network bandwidth between network components (MCD platforms, layer 2 switches, layer 3 switches, or routers).	Configure Bandwidth Management zones. Refer to the *MCD Engineering Guidelines* and System Administration Tool online help for details.
		Add bandwidth.
	Packet loss or bit errors occurring due to incorrect Quality of Service settings.	Ensure Quality of Service setting are correct on all network components (MCD platforms, layer 2 switches, layer 3 switches, routers, and VLANs). Refer to the *MCD Engineering Guidelines* for QoS guidelines.
Conference calls or calls that use complex routing	Excessive speech transcoding.	Simplify the network. Configure the network to use only G.711 transmission format.
IP or analog calls across Loop Start trunks	Analog trunk descriptor set incorrectly.	Run the Line Measurement Tool (see page 56) to apply correct settings.
Calls involving analog phones	ONS/OPS station descriptors set incorrectly.	In the MCD System Administration Tool, access the **ONS/OPS Descriptor** form and check that the signaling and timing parameters are set correctly. Refer to the form's online help for details.
Calls involving speakerphones	One of the parties on the call is speaking too loudly.	Reduce volume or switch to handset mode.
Calls involving a specific destination	One of the parties is using a non-TIA compliant phone (typically a far-end phone outside the customer's network).	Refer to "Identifying the Source of Echo" on page 55 to confirm that the echo is originating in the far-end equipment. Inform customer that this issue is not being caused by Mitel equipment.
All calls involving a specific phone.	Faulty microphone in handset, headset, or speakerphone.	Replace phone or headset.
All calls involving a wireless handset or headset	Low battery power in wireless handsets or headsets or user is out of range of base station.	Charge phone batteries. Stay within range of base stations Ensure base station coverage is adequate. Refer to the wireless system site survey documentation for details.

Distorted Music On-Hold

Table 18: Distorted Music On-Hold

The issue is occurring . . .	Probable Cause	Corrective Action
Constantly on embedded Music-on-hold audio	Music file format is incorrect.	Replace music file with a file that has the following format: • *Audio type*: Microsoft Windows WAV (.wav) • *Audio format*: CCITT µ-Law or A-Law 8.000 kHz, 8 Bit, Mono Refer to the System Administration Tool online help for additional details.
	Incorrect music file format for geographical location.	North American systems require µ-law; European systems require A-law.
Constantly on analog music-on-hold audio.	Volume of music source is set too high.	Reduce volume of Music On Hold source.
Intermittently on IP phones while music on hold (background music) is playing.	Too many transcoding hops.For example, audio is transcoded from G711 to G729 and back to G711.	None.

Crosstalk

Table 19: Crosstalk

The issue is occurring on . . .	Probable Cause	Corrective Action
Multiple calls involving analog phones	Faulty telephone cables or wiring connections (either on premise or outside plant). For example, audio signals jumping between frayed cables at cross connections, or water in outdoor cabling allowing signals to cross between the wires.	Check wiring connections, cabling, and grounding.
Multiple calls involving analog phones on the same Analog Service Unit (ASU), Analog Main Board (AMB), or Analog Options Board (AOB).	TDM fabric (switching matrix) is faulty.	Replace ASU, AMB, or AOB. Refer to the *MCD Technician's Handbook* for instructions on how to replace an AMB or AOB.

Dropped Calls

Table 20: Dropped Calls

The issue is occurring on . . .	Probable Cause	Corrective Action
Calls involving IP phones on LAN or WAN.	Lack of bandwidth, extreme packet loss, or excessive network congestion.	Add bandwidth.
	Bandwidth Management and Call Admission Control is configured incorrectly.	Check Bandwidth Management zone configuration. Refer to the *MCD Engineering Guidelines* and System Administration Tool online help for details.
	Packet loss or bit errors occurring due to incorrect Quality of Service settings.	Ensure Quality of Service settings are correct on all network components (MCD platforms, layer 2 switches, layer 3 switches, routers, and VLANs). Refer to the *MCD Engineering Guidelines* for QoS guidelines.
Calls involving IP phones on a specific MCD platform, layer 2 switch, layer 3 switch, or router.	Layer 2 or layer 3 switch QoS settings are incorrect for signalling packets.	Ensure Quality of Service settings are correct on all network components (MCD platforms, layer 2 switches, layer 3 switches, routers, and VLANs). Refer to the *MCD Engineering Guidelines* for QoS guidelines.
IP to IP calls (no audio path) between IP phones on a specific MCD platform.	Ensure E2T card in MCD platform has booted up.	In the System Administration Tool, access the **Maintenance Commands** form and enter the "DBMS Save" command to restart the E2T card.
	Programming error. Hotel/Motel Call Blocking option is enabled and preventing calls.	In the MCD System Administration Tool, access the **Hotel Options Assignment** form. Ensure that Call Blocking is not enabled.

Silence

Table 21: Silence

The issue is occurring on . . .	Probable Cause	Corrective Action
One end of the call or both ends of the call.	Incorrect firewall settings.	Ensure that ports are configured correctly in the MCD platform Port Forward Table to allow external traffic to reach resources on the internal network. Refer to the *MCD Engineering Guidelines* for firewall details and a complete list of the MCD ports.
One end of the call only (multiple IP phones).	Default MCD gateway IP address not configured correctly in MCD platform.	In the MCD System Administration Tool, access the **System IP Properties** form and ensure that the Gateway IP address is entered correctly.

Table 21: Silence

The issue is occurring on . . .	Probable Cause	Corrective Action
One end of the call only (single IP phones).	Default gateway IP address or subnet mask not configured correctly in IP Phone.	Check Gateway IP address of the IP phone. Access the Configuration (Debug) Menu on the phone to view its settings and network parameters, including its Gateway IP address (refer to the *MCD Troubleshooting Guide* for instructions).
Silence occurs during an IP phone call.	IT administrator was performing maintenance on routers during business hours. Router sent a redirect signal to either the IP phone or MCD platform while the call was in progress. The redirect signal caused the connection to be broken.	Perform router maintenance after business hours.

Tinny Audio (Drain Pipe)

Table 22: Tinny Audio (Drain Pipe)

The issue is occurring on . . .	Probable Cause	Corrective Action
Calls involving the VoIP network.	Call is using compression.	Configure Bandwidth Management and Call Admission Control to ensure adequate bandwidth. See Bandwidth Management in the MCD System Administration Tool online help for details.
		Remove compression and add bandwidth.
IP or TDM calls over digital trunks.	Faulty digital link descriptor programming.	Access the **Digital Link Descriptor** form in the System Administration tool. Ensure that the link descriptors are programmed correctly.
Calls with a specific external phone.	External party is using a phone that is not TIA-compliant (for example: a cell phone or a residential wireless phone).	Refer to “Identifying the Source of Echo” on page 55 to confirm that the echo is originating in the far-end equipment. Inform customer that this issue is not being caused by Mitel equipment.
Calls involving a speakerphone.	One party is using a speakerphone.	Reduce volume of speakerphone.
		Relocate speakerphone away from reflective surfaces.
		Instruct user to move closer to speakerphone.
		Switch to handset.

Delayed Response Time (Citizen's Band Effect)

Table 23: Delayed Response Time (Citizen's Band Effect)

The issue is occurring on . . .	Probable Cause	Corrective Action
Calls involving IP phones.	Network latency.	Add bandwidth. Ensure network latency does not exceed the limits that are specified in the *MCD Engineering Guidelines*.
		Configure Bandwidth Management zones. Refer to the *MCD Engineering Guidelines* and System Administration Tool online help for details.
	Packet loss or bit errors occurring due to incorrect Quality of Service settings.	Ensure Quality of Service settings are correct on all network components (MCD platforms, layer 2 switches, layer 3 switches, routers, and VLANs). Refer to the *MCD Engineering Guidelines* for QoS guidelines.
Calls involving IP phones on a specific MCD platform, layer 2 switch, layer 3 switch, or router.	Quality of Service settings are incorrect, resulting in a packet bottleneck.	Ensure Quality of Service settings are correct on all network components (MCD platforms, layer 2 switches, layer 3 switches, routers, and VLANs). Refer to the MCD Engineering Guidelines for QoS guidelines.
Some long-distance calls.	Call occurs over a satellite link that is introducing delays.	Refer to "Identifying the Source of Echo" on page 55 to confirm that the echo is originating in the far-end equipment. Inform customer that this issue is not being caused by Mitel equipment.
Calls that use complex routing or conference calls	Excessive speech transcoding.	Prevent complex call routing by simplifying the network. Configure the network to use only G.711 transmission format.
Calls involving a Teleworker	Quality of Service is not supported by the Teleworker's Internet Service Provider (ISP).	Subscribe to a different ISP that supports QoS. If you are planning to provide an application, such as Teleworker or Blackberry MVS solution over a non-QoS network, it is important to inform the customer of the potential quality issues, prior to implementing the service.

Low or High Audio

Table 24: Low or High Audio

<table>
<tr><th>The issue is occurring on . . .</th><th>Probable Cause</th><th>Corrective Action</th></tr>
<tr><td rowspan="3">Calls over Loop Start trunks</td><td>Incorrect Loop Start trunk descriptor settings.</td><td>Run the Line Measurement Tool (see page 56) to apply correct settings.</td></tr>
<tr><td>Incorrect Loss/Level plan is selected on MCD platform.</td><td>The Loss/Level plan is determined by the Country setting in the Licensing and Options Selection form of the MCD platform. Ensure that the Country setting is correct. Refer to the MCD Hardware Technical Reference Manual for the supported Loss/Level plans.</td></tr>
<tr><td>Third-party platform is using a non-compliant loss plan.</td><td rowspan="2">Refer to “Identifying the Source of Echo” on page 55 to confirm that the echo is originating in the far-end equipment. Inform customer that this issue is not being caused by Mitel equipment.</td></tr>
<tr><td>Calls involving a specific external phone</td><td>External party is using a phone that is not TIA-compliant (for example: a cell phone or a residential wireless phone).</td></tr>
</table>

Using Tests and Diagnostic Tools

If necessary, you can use the tests and diagnostic tools described in this section to collect additional data on specific voice quality issues.

Identifying the Source of Echo

It is normal to receive reflection on a call from the PSTN to a phone on the MCD system.

- If the echo delay looking into the PSTN, measured at the demarcation point, is short (< 50 ms), then the PSTN and CPE at the other end of the connection are not responsible for canceling the echo. In this case Mitel equipment must cancel the echo from the PSTN, to prevent the echo from being heard by the IP phones. Without cancellation, the IP phones would hear echo because of the delay over the IP connection. If the delay between the demarcation point and phone is short, which would be the case if a Mitel analog phone is used, then there is no need for Mitel equipment to cancel the echo.
- If the echo delay looking into the PSTN, measured at the demarcation point, is long (> 50 ms), then the cause is likely the long delay audio path on the customer premise equipment on the other side of the PSTN. In this case, the customer premise equipment is responsible for removing the reflection from the long delay path before it enters the PSTN (short delay path).

If a Mitel IP phone is experiencing echo on an external call, you can identify the source of echo by placing a call from a Mitel analog phone to the same external destination:

- If you hear echo, the customer premise equipment on the other side of the PSTN is the source. The echo must be corrected in the CPE on the other side of the PSTN.
- If you do not hear echo when you call from a Mitel analog phone, the MCD equipment is likely the source and the echo must be corrected in the Mitel equipment.

Figure 10 illustrates these two possibilities:

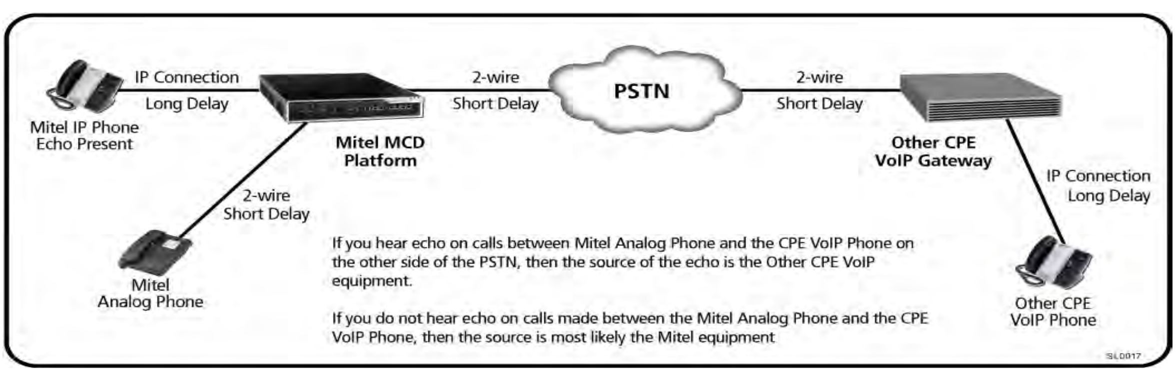

Figure 10: Identifying the Source of Echo

Line Measurement Tool

Use this tool to determine the correct line settings for Loop Start (LS) trunks that are connected to the AX Controller Card Chassis, Analog Main Board, Analog Option Board, or Analog Service Unit. You can eliminate most trunk-related echo problems by applying the correct line settings in the MCD trunk forms.

You can configure the Line Measurement Tool to automatically apply the line settings required to remove nearly all of the reflection caused by hybrids on the local 2-wire loop. To have the tool automatically apply the correct settings, enable the **Automatic Settings option** in the **Analog Trunks** form of the trunk before running the tool (see Step 8 of "Configure System for Testing using Loopback" on page 57).

The following tests are supported:

- **Individual or Batch Line Quality Test**: Reports the loss level, Echo Return Loss (ERL), and line quality for a specified trunk(s). A recommended Balance Network Setting and Trunk Category is provided for each LS trunk, based on the test results.
- **Individual or Batch Distortion/Echo Test**: Measures the non-linear distortion effects (for example, from clipping) of a specified LS trunk(s) for each candidate balance circuit setting. Run the Distortion/Echo test only if Mitel Technical Support requests the test results.

Before You Begin

The Line Measurement Tool is supported on all SHARC DSPs with the exception of the ADSP 21161. To determine what card is installed in your system, check the **Hardware Modules** form under the **Card Description** column heading.

Before running these tests, ensure that you have

- ❒ determined if your CO provides a trunk with milliwatt tone or silence tone. If not, you must run the tests in loopback mode.
- ❒ configured the trunks for loopback testing if required (see below).
- ❒ determined if all LS trunks are from the same CO. For loopback, tests must be performed on LS trunks that are provided from the same CO into the MCD.
- ❒ established a connection between the LS trunk line on the AX Controller card chassis, Analog Main Board, Analog Option Board, or ASU II and the local CO.
- ❒ obtained the milliwatt tone number from the CO.
- ❒ obtained the silence tone number from the CO.
- ❒ recorded the physical location identifier (PLIDs) of the LS trunks.
- ❒ before you can run these tests on a trunk, the trunk must be idle. If the trunk is in use and you attempt to run the test, you will receive an error message. While a test is in progress, the system cannot use the trunk to make calls. You cannot run these tests on trunks that have been busied out.

You can run the Loop Start (LS) measurement tests using calls between an MCD and CO, or by looping calls through one LS trunk on the MCD back through other LS trunk on the same MCD.

Configure System for Testing using Central Office Numbers

To run the Loop Start (LS) measurement tests, place calls between an MCD and a Central Office:

1. Program the following numbers in the Trunk Circuit Descriptor - CO:
   - Central Office Milliwatt Telephone Number
   - Central Office Silence Telephone Number
2. To have system automatically apply the recommended settings from the results of the Line Quality Tests to the trunks, enable the **Auto Select** option in the **Analog Trunks** form. If you don't select this option, you must manually program the settings into the fields of the **Analog Trunks** form.
3. Configure the system to automatically email the System Diagnostic Report results from the Distortion/Echo test to Mitel Technical Support for analysis.

Configure System for Testing using Loopback

You can also run the Line Loss Level test on an LS trunk by setting up loopback through another LS trunk on the MCD. You must program the trunks and designate one of the trunks as an answer point for the milliwatt tone.

Caution: Do not change the settings of the Trunk Circuit Descriptor until you have tested all the trunks.

To set up loopback:

1. Record the CO phone number for each trunk.
2. Program the LS trunks using the default (normal) settings for your region.
3. Program Automatic Route Selection (ARS) to allow outgoing calls.
4. System **Access Points** form.
   - Program an unused directory number for the Milliwatt Test Directory Number.
5. **CO Trunk Circuit Descriptor** form.
   - Ensure that the trunk used for loopback testing is programmed with a separate Trunk Circuit Descriptor during testing.
   - Set the Trunk Category to **Short**.
6. **Trunk Attributes** form.
   - Program the Milliwatt Test Directory Number as an answer point for the trunks.
   - Select the trunk service number of the incoming LS trunk.
   - Set the following answer points to the Milliwatt Test Directory Number that you programmed in the previous step:
     Non-dial In Trunks Answer Point - Day
     Non-dial In Trunks Answer Point - Night 1, and
     Non-dial In Trunks Answer Point - Night 2

7. Program the following numbers in the **Analog Trunks** form:
   - In the Milliwatt Test Trunk Number field, enter the trunk index number of the trunk that will provide the Milliwatt tone to the trunk under test.
   - In the CO Telephone Number field, enter the CO telephone number (maximum 26 digits) of the LS trunk that you want to test.
8. To have the recommended settings from the results of the Line Quality Test applied automatically to a trunk, enable the **Auto Select** option in the **Analog Trunks** form of the trunk.
9. Make a test call using ARS, or from an external phone, to one of the trunks and verify that milliwatt tone is received.

Individual or Batch Line Quality Test

Run these tests after business hours during periods of low call traffic.

To run the tests:

1. Ensure that you have configured the system to support testing. See
   - "Configure System for Testing using Central Office Numbers" on page 57, or
   - "Configure System for Testing using Loopback" on page 57.
2. Access the **Line Quality Measurement** form.
   - Click **Run Test**.
   - Select the Test Type:

     -To test a single trunk select Individual Line, then select the PLID of the trunk.

     -To run the test on all the eligible LS trunks in the system, select Batch Line. The PLID field is disabled. The test may take up to 12 minutes to return results depending on the number of configured lines.
3. Click **Run**.
4. If the test is successful, the test results are displayed in the bottom frame of the form.
5. Enter the recommended Balanced Network Setting and Trunk Category in the **Analog Trunks** form. If you previously set Auto Select to **Yes** in the **Analog Trunks** form, the system automatically updates these fields with the recommended settings.

Individual or Batch Distortion/Echo Test

This test takes approximately 15 minutes to complete on each trunk. While the test is running on a trunk, the system takes the trunk out of service.

Caution: Only run this test at the request of Mitel Technical Support. System performance could be degraded while the test is running.

To run the tests

1. Ensure that you have configured the system to support testing using Central Office numbers or Loopback. See
   - "Configure System for Testing using Central Office Numbers" on page 57, or
   - "Configure System for Testing using Loopback" on page 57.

2. Access the **Line Measure Quality** form.
3. Click **Run Test**.
4. Select the Test Type. To test a single trunk select **Individual Distortion/Echo** then select the PLID of the trunk. To run the test on all the eligible LS trunks in the system, select **Batch Distortion/Echo**. The PLID field is disabled.
5. Click **Run**.
6. If the test is successful, the test results are saved in the "/db/database/LSMeasure" directory.
7. Run a System Diagnostic Report to automatically collect all the echo/distortion files. Then, send the System Diagnostic Report files to Mitel Technical Support for analysis.

Voice Over IP Troubleshooting Web Site

Access the following web site for information on how to troubleshoot VoIP problems, free online diagnostic tools, and useful information on voice quality and packet statistics:

http://www.voiptroubleshooter.com/about/index.html

Network Traffic Analyzer

Use a network protocol analyzer (for example Wireshark®) to capture and view the traffic running on the IP network. You can use a network traffic analyzer to

- capture data "from the wire" on a live network connection or read from a previously captured file that contains recorded packets.
- read live data from different types of networks
- browse and filter displayed network data
- detect VoIP calls in the captured data
- verify Quality of Service settings on the MCD platform, layer 2 Switches, and routers
- listen to problematic calls.

> **Note:** Before using a network traffic analyzer, such as Wireshark, ensure that you disable voice encryption in the **System Options Assignment** form of the MCD System Administration Tool. After you have finished, using the analyzer, re-enable voice encryption.

You can download the Wireshark tool from the following site: http://www.wireshark.org/. This site also provides instructional videos on how to use Wireshark. When capturing the call using WireShark, you should also record the call using a handset audio tap tool. It is very common to receive reports of voice quality problems and have traces that do not show any problems. This could be because the

- audio problems were introduced downstream from the IP network
- problems are sporadic and were not captured, or

- users expectations of call quality are unrealistic.

The handset audio tap recordings will help you identify these cases and save time and effort.

Handset Audio Tap

Audio problems can be introduced at any point in the audio path between the parties in the call. Wireshark and other monitoring tools do not necessarily monitor what the caller actually hears because the audio can be modified by down stream devices. To record what is actually heard by a caller, you need a handset audio tap such as the "THAT-2 Telephone Handset Audio Tap" (http://www.jkaudio.com/that-2.htm).

You insert the handset tap into the handset cord and then record the audio using a PC sound card or other recording device. You should also get a Wireshark trace of the call while it is being recorded with the handset audio tap.

VoIP Bandwidth Management on MCD Platforms

Quality issues on VoIP calls can be caused by a lack of bandwidth. The Bandwidth Management feature in the MCD platform allows you to track and manage bandwidth consumption by the VoIP media streams. The bandwidth used by the media stream is measured between zones of co-located IP phones. At predetermined bottleneck points in the network, it performs the following functions for the voice data packets:

- Measures and reports consumed and available bandwidth
- Generates maintenance alarms when bandwidth consumption exceeds configured threshold levels.
- Provides Call Admission Control, that is, the rejection of new calls through a specific bottleneck point when consumed bandwidth exceeds maximum configured levels.

Use bandwidth statistics and alarm thresholds to ensure that adequate bandwidth is allocated between devices in the network. If Bandwidth Management alarms are occurring between zones on a regular basis, you should increase the bandwidth available between the zones. On the other hand, if alarms are never generated between two zones, the Bandwidth limit for the media stream between those two zones may be set too high. Some of the bandwidth could possibly be reallocated to data requirements.

You use the following MCD System Administration Tool forms to manage bandwidth:

- Bandwidth Management Configuration
- 15-Minute Historical Bandwidth Statistics
- Current Bandwidth Statistics.

Refer to the System Administration Tool online help for instructions.

Internet Speed Test

Test your internet upload and download speeds from a PC on the LAN using a speed test tool. You can access a speed test tool at the following URL: http://www.bandwidthplace.com/.

A slow internet connection (lack of bandwidth) can cause voice quality issues on SIP calls across the wide area network (WAN). You require 80 to 90 kbps of upload and download bandwidth for a single two-way VoIP conversation using G711 codec. Contact your Internet Service Provider (ISP) if you are not receiving your full level of internet service. If your level of service is not adequate to support calls across the WAN, you may need to upgrade your service.

TRACERT Maintenance Command

This MCD System Maintenance Command traces the path

- from the Real Time Complex card in the MCD platform to a destination IP address, or
- from IP sets to a destination IP address.

The system response identifies the IP address of routers (or hops) traversed and the time it took to make three round trips to each device (including the destination) and back. TRACERT is useful for determining where performance bottlenecks might be occurring.

Note that the TRACERT command supports the following sets only:

- 5212 IP
- 5215 Dual Mode
- 5220 Dual Mode
- 5224 IP
- 5235 IP
- 5312 IP
- 5320 IP
- 5324 IP
- 5330 IP
- 5340 IP
- 5360 IP
- Navigator
- Telematrix 3000IP

Enter this command in the Maintenance Commands form of the MCD System Administration Tool using the following format:

TRACERT

Qualifiers:

RTC <ip address> /HOPS n /WAIT n

IPSET <ip address> TO <ip address> /HOPS n /WAIT n

Refer to the TRACERT Maintenance Command in the System Administration Tool online help for details.

Ping Test

You can use a ping utility to determine the round-trip delay between devices on the network.

MCD System Administration Tool Ping Utility

The **IP Telephones** form in the MCD System Administration tool provides a ping utility that you can use to determine whether a programmed, in-service IP telephone is accessible. This ping utility is not supported for SIP devices.

Windows and Linux Ping Utilities

To perform a ping test from a PC running Microsoft Windows XP:

1. Obtain the IP address of the IP phone that is experiencing voice quality issues. You can access the Configuration (Debug) Menu on the phone to view its settings and network parameters, including its IP address (refer to the MCD Troubleshooting Guide for instructions).

Note: A Mitel IP phone can be configured with a static IP address or it can obtain its IP address from a DHCP server. If the IP phone gets its address from a DHCP server, the IP address can change whenever the IP phone reboots.

Note: The Configuration (Debug) Menu on Mitel IP phones provides a ping test. However, it only tests connectivity and does not indicate the delay time.

2. Go to a PC that is on the same subnet as the IP phone that is experiencing voice quality issues.
3. From the Windows **Start** menu, click **Run**, and type "cmd" and then click **OK** to open a command window.
4. At the prompt, type "ping" followed by the IP address of the Mitel IP phone and press **Enter**. Figure 11 shows an example:

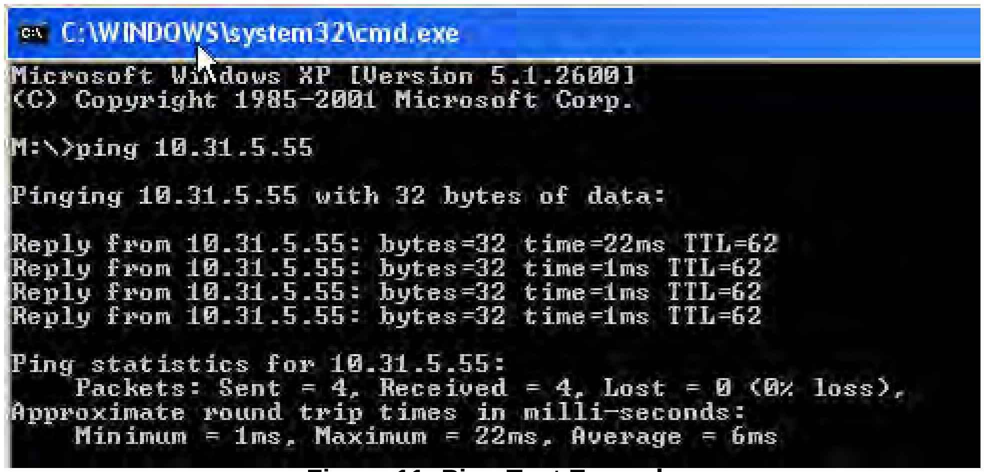

Figure 11: Ping Test Example

5. Divide the maximum round trip time in half to obtain the end-to-end delay time. If the end-to-end delay time exceeds 80 ms, delay is likely a cause of any voice quality issues.

Voice Quality Statistics for Mitel IP Phones

You can view jitter, packet loss, and latency (delay) statistics for Mitel IP phones in the **Voice Quality Statistics** form in the MCD System Administration Tool. The form also lists other useful information such as the device type of the endpoint, the last CODEC used in the call, the IP address of the subnet and the IP address of the endpoint.

The following forms in the MCD System Administration tool support voice quality statistics:

- Voice Quality Monitoring
- User Authorization Profiles
- Voice Quality Statistics.

To view the voice quality statistics for a call from an IP phone:

1. Log into the MCD System Administration Tool.
2. In the **Voice Quality Monitoring** form, ensure that **Voice Quality Monitoring** is set to **True**.
3. In the **User Authorization Profiles** form ensure that the User Profile associated with the Enterprise Manager has "Application" enabled in the Access field. This allows access to the MiXML management.
4. Place a call from the IP phone that is experiencing the voice quality issue. If the problem is intermittent, ensure that you view the statistics while the problem is present on the call. Because the data network is dynamic, you may not see evidence of a voice quality issue if you check the statistics during favorable network conditions (for example, during a period of low traffic).

Note: Voice Quality statistics are only supported for the following phones: 5360 5340, 5330, 5324, 5320, 5312, 5304, 5235, 5224, 5220 dual mode, Navigator, TeleMatrix 3000IP, 5560 IPT (Master and slave devices are displayed separately), 5215 dual mode 5212. Refer to the System Administration Tool online help for the most up-to-date list of supported phones. Note that SIP phones, including Mitel SIP phones, do not support Voice Quality Statistics.

5. In the **Voice Quality Statistics** form, search for the IP phone that is experiencing voice quality issues by directory number, IP Address, or MAC Address. You can obtain this information for a phone from the **IP Telephones - Programmed** form.
6. Check the values for the following fields:

Field	Description	Interpreting the data
Packets Received	The total number of packets received during a report interval.	Calculate the percentage of packet loss using the following formula: $\left(\frac{\text{\# of packets lost}}{\text{\# of packets lost + \# of packets received}}\right) \times 100$ If the percentage is greater than 2 %, then packet loss is causing the voice quality issue.
Packet Loss	The total number of packets lost during a call.	
Average Jitter (ms)	Jitter is the variation in delay (ms).	If this value is greater than 60 ms, Jitter is a factor in causing the voice quality issue.
Delay (ms)	The total time required for a packet to travel from the IP phone to the other IP phone.	If this value is greater than 80 ms, Delay is a factor in causing the voice quality issue.

System Diagnostic Reports

The MCD system records diagnostic information about the system performance in a series of report files. This diagnostic information includes xrtc, pstswlog.db, hdrwswlog.db logs, call control statistics. The information collected in these files is intended to be used by Mitel Technical Support to diagnose system problems. Use the **System Diagnostics Reporting** form in the MCD System Administration Tool to collect these files.

Note: Run system diagnostics only during periods of low traffic. If you run system diagnostics during periods of high traffic, the system could drop calls on IP trunks.

Station Message Detail Recording (SMDR) Logs

Review the MCD System Administration Tool SMDR call logs for information on problem calls:

- call start time
- calling party
- called party

- call duration
- digits dialed (maximum 26 digits)
- outgoing and incoming trunk numbers - external SMDR only
- conferences and transfers
- trunks involved in route optimization - external SMDR only.

Refer to System Administration Tool online help for instructions.

Appendix: Additional Self-Help Resources

Mitel Communications Director Troubleshooting Guide

For general troubleshooting information, refer to the *MCD Troubleshooting Guide*. This guide is available from the Mitel Customer Documentation site on Mitel Online.

To access this guide:

1. Log into Mitel OnLine.
2. From Mitel OnLine, click **Technical** and then click **Product Documentation**.
3. Under **Communications Platforms**, click Mitel Communications Director and then click the platform (for example, Mitel MCD).
4. Click the Troubleshooting Guide link.

Checking the Mitel Knowledge Base

The Mitel Knowledge Base is your source for product information covering a large range of technical support methods. All new Technical Bulletins (TBs), Release Notes (RNs), How to Articles and Troubleshooting Articles are posted in the Knowledge Base. Access to the Knowledge Base is given with access to Mitel Online (MOL).

To access the Knowledge Base:

1. Log into Mitel OnLine.
2. From Mitel OnLine, click **Technical** and then click **Knowledge Base**.
3. Select the product name (for example, 3300 Integrated Communications Platform).
4. Select the type of article you are looking for or choose **All**.
5. Click **Search**.

Using the Online Service Desk (OSD)

The Online Service Desk provides up-to-date information on your Support Notifications and Technical Credit balance. This portal lets you view the status of your open tickets, update Support Notifications with new information for our technical support team, and check your technical credit usage. Once the issue has been resolved, your ticket will be updated. OSD will automatically display all active support notifications when you log in.

To access the Online Service Desk:

1. Log into Mitel OnLine.
2. From Mitel OnLine, click **Support**, then click **Online Service Desk**. The first time you log in you will be asked for your company’s Technical Support ID.

3. Choose your channel partner location.
4. View your open and closed tickets.
5. Update your open tickets with any new information and the Mitel technical support specialist will be notified of your updates by email.

Note: Online Service Desk (OSD) response time is three business days. The OSD should not be used to obtain emergency support services.

6. Display a summary of your technical credits (depending on your access privileges).
7. Ensure that you enable your profile to allow Mitel to notify you via email when your ticket is updated.

Checking for Fixes in Software Patches

Before calling technical support with a problem, check to see if there is a software update available on Mitel Online that includes a fix for the problem.

- Major software patches are provided on Mitel Online.
- Minor software updates are available from Technical Support via a Mitel FTP site. The Known Product Issues (KPI) document (MKB article 08-5104-00051) is a summary of 3300 field found problems fixed from release 7 to the latest release.

If an update is available with the required fix, you can download the software patch or update and install it on the system.

Fixed in Latest Software Patch?

To determine if a fix for your problem is available in the latest software patch:

1. Log into Mitel OnLine.
2. Click click **Technical** and then click **Software Downloads**.
3. Click the product name (for example, 3300 Integrated Communications Platform).
4. Click the latest software update.
5. Open the Release Notes (RN).
6. In the RN, review the list of problems (Defects) that are resolved in this software update. If your required fix is listed, read the RN carefully to ensure that the update is appropriate, download the update, and then install it on the system. See the *Technician's Handbook* for software installation instructions.
7. Review the Known Product Issues (KPI) document (MKB article 08-5104-00051) available on Mitel Online.

Fixed in Previous Software Patch?

If the software version on your system is a few releases old, the fix may have been provided in a previous software patch. To check the problems that have been fixed in previous software patches

1. Log into Mitel OnLine.
2. From Mitel OnLine, click **Technical**, and then click **Knowledge Base**.
3. Select the product name (for example, 3300 Integrated Communications Platform).
4. Set the Article Type to Release Notes (RN) and then click **Start Search**.
5. Open the RNs that have been issued since the release of your system's current software and check the list of problems (Defects) that have been fixed.
6. If you find a fix for your problem, download and install the latest software patch. See the *Technician's Handbook* for software installation instructions.

Fixed in Software Update?

To determine if a problem has been fixed in a minor software update:

1. Log into Mitel OnLine.
2. From Mitel OnLine, click **Technical**, and then click **Knowledge Base**.
3. Select the product name (for example, 3300 Integrated Communications Platform).
4. Enter "KPI" and then click **Start Search**.
5. Open and review the Known Product Issues (KPI) fixed lists.
6. If you find a fix for your problem, contact Technical Support to obtain the software update.

Index

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S

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V

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